Chapter 1

Quality-of-Service Scheduling for WiMAX Networks

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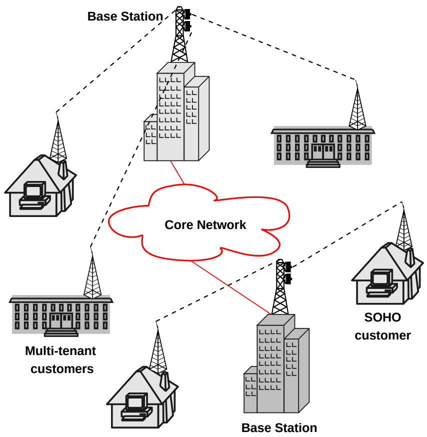
The broadband wireless world is moving towards the adoption of WiMAX (the commercial name of the IEEE 802.16 standard) as the standard for broadband wireless Internet access. This will open up a very large market for industry and operators, with a major impact on the way Internet access is conceived today. On the other hand, the emergence of innovative multimedia broadband services is going to impose severe Quality-of-Service (QoS) constraints on underlying network technologies. In this work, after a brief review of the IEEE 802.16 standard, we intend to present an in-depth discussion of its QoS support features. We point out the scheduling algorithm as the critical point in QoS provisioning over such networks, and discuss architectural and algorithmic solutions for an efficient support of multimedia flows. Performance measurements obtained from an experimental testbed are also presented. The paper concludes with a description of the key research challenges in the area, and provides a roadmap for the research in the field.

1.1. Introduction

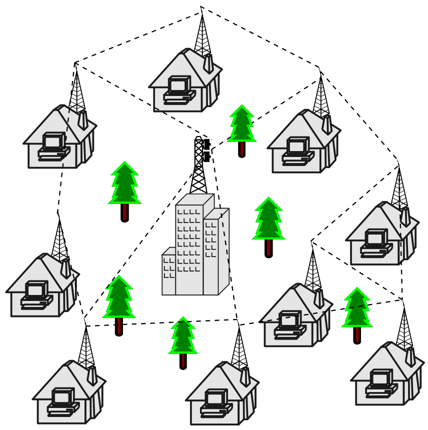
The IEEE 802.16 standard,[1] promoted by the WiMAX (Worldwide Interoperability for Microwave Access) forum,[2] will be the leading technology for the wireless provisioning of broadband services in wide area networks. Such technology is going to have a deep impact on the way Internet access is conceived, by providing an effective wireless solution for the last mile problem.

The market for conventional last mile solutions (e.g., cable, fiber etc.) presents indeed high entrance barriers, and it is thus difficult for new operators to make their way into the field. This is due to the extremely high impact of labor-intensive tasks (i.e., digging up the streets, stringing cables etc.) that are required to put the necessary infrastructure in to place. On the other hand, the market is experiencing an increasing demand for broadband multimedia services,[3] pushing towards the adoption of broadband access technologies. In such a situation, Broadband Wireless Access (BWA) represents an economically viable solution to provide Internet access to a large number of clients, thanks to its infrastructure-light architecture, which makes it easy to deploy services where and when it is needed. Furthermore, the adoption of ad hoc features, such as self-configuration capabilities in the Subscriber Stations (SSs) would make it possible to install customer premises equipment without the intervention of a specialized technician, so boosting the economical attractiveness of WiMAX-based solutions. In this context, WiMAX is expected to be the key technology for enabling the delivery of high-speed services to the end users.

Typical BWA deployments will rely on a Point-to-MultiPoint (PMP) architecture, as depicted in Fig. 1.1(a), consisting of a single Base Station (BS) wirelessly interconnecting several SSs to an Internet gateway. The standard also supports, at least in principle, mesh-based architectures, like the one plotted in Fig. 1.1(b). While WiMAX-based mesh deployments could play a relevant role in the success of such technology, the current standard[1] is far from offering a real support to such architecture. Therefore, we intend to restrict the scope of our work to the PMP architecture only.



(a) Point-to-Multipoint



(b) Mesh

Fig. 1.1. Typical WiMAX system configuration.

In terms of raw performance, WiMAX technology is able to achieve data rates up to 75 Mb/s with a 20 MHz channel in ideal propagation conditions.[4] But regulators will often allow only smaller channels (10 MHz or less) reducing the maximum bandwidth. Even though 50 km distance is achievable under optimal conditions and with a reduced data rate (a few Mb/s), the typical coverage will be around

5 km in non-line-of-sight conditions and around 15 km with an external antenna in a line-of-sight situation. Moreover, such a wide coverage makes it possible, and economically viable to provide broadband connectivity in rural and remote areas, a market which is usually not covered by traditional service providers.

The fundamental requirements for WiMAX to define itself as a possible winning technology are data reliability and the ability to deliver multimedia contents. Indeed, the provision of QoS guarantees will be a pressing need in the next generation of Internet, in order to enable the introduction of novel broadband multimedia applications. Users are actually getting more and more interested in broadband applications (e.g., video streaming, video conferencing, online gaming etc.) that require assurances in terms of throughput, packet delay and jitter, in order to perform well. This applies also to WiMAX networks, which have also to face all the problems related to the hostile wireless environment, where time-varying channels and power emission mask constraints make it difficult to provide hard QoS guarantees. This entails the definition of a medium access control protocol which is able to effectively support such multimedia applications while, on the other hand, it efficiently exploits the available radio resources. The IEEE 802.16 standard encompasses four classes of services, with different QoS requirements and provides the basic signalling between the BS and the SSs to support service requests/grants. However, the scheduling algorithms to be employed in the BS and the SSs are not specified and are left open for the manufacturers to compete.

In this paper, after a brief review of the standard fundamentals, we will provide an in-depth overview and discussion on the QoS support provided by WiMAX technology. Particular attention will be devoted to scheduling algorithms for WiMAX networks. We will survey the existing literature, and point out some common issues involved in well-known technologies (e.g., wireless ATM), from which a system designer can draw to design an efficient scheduler without starting from scratch. Performance measurements obtained from an experimental test-bed are also presented. The paper concludes with an overview of the actual research challenges, pointing out and detailing the most promising directions to pursue for research in this field.

1.2. WiMAX Technology Overview

WiMAX is the commercial name of products compliant with the IEEE 802.16 standard. Effectively replicating the successful history of IEEE 802.11 and Wi-Fi, an industrial organization, the WiMAX Forum has been set up to promote the adoption of such technology and to ensure interoperability among equipment of different vendors. This forum, which includes all the major industrial leaders in the telecommunication field, is expected to play a major role in fostering the adoption of IEEE 802.16 as the *de facto* standard for BWA technology.

The general protocol architecture of the IEEE 802.16 standard is depicted in

Fig. 1.2. As can be seen, a common media access control (MAC) is provided to work on top of different physical layers (PHY). The interface between the different PHYs and the MAC is accomodated as a separate sublayer, the transmission convergence sublayer. A Convergence Sublayer (CS) is provided on top of the MAC, to accomodate both IP as well as ATM-based network technologies. A basic privacy support is provided at the MAC layer.

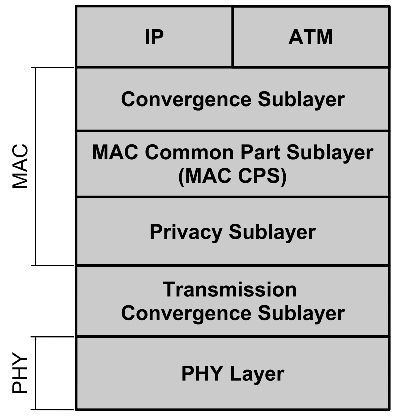


Fig. 1.2. IEEE 802.16 protocol architecture.

In its first release in 2001, the 802.16 standard addressed applications for a static scenario in licensed frequency bands in the range between 10 and 66 GHz, where the use of directional antennas are mandatory to obtain satisfactory performance figures. In a metropolitan sub-area, however, line-of-sight operations cannot be ensured due to the presence of obstacles, buildings, foliage etc. Hence, subsequent amendments to the standard (802.16a and 802.16-2004) have extended the 802.16 air interface to non-line-of-sight applications in licensed and unlicensed bands in the $2-11$ GHz frequency band. With the revision of IEEE standard document 802.16e, also some mobility support will be provided. Revision 802.16f is intended to improve multi-hop functionality, and 802.16g is supposed to deal with efficient handover and improved QoS.

WiMAX technology can reach a theoretical 50 Km coverage radius and achieve data rates up to 75 Mb/s,[4] although actual IEEE 802.16 equipments are still far from these performance figures. As an example, in[5] the authors report the outcomes of some bit-level numerical simulations performed assuming a channel width of 5 MHz and a Multiple-Input Multiple-Output (MIMO) 2x2 system (which reflects the most common actual equipment), showing that, under ideal channel conditions, data rates up to 18 Mb/s can be attained.

Duplexing is provided by means of either Time Division Duplexing (TDD) or Frequency Division Duplexing (FDD). In TDD, the frame is divided into two subframes, devoted to downlink and uplink, respectively. A Time-Division Multiple

Access (TDMA) technique is used in the uplink subframe, the BS being in charge of assigning bandwidth to the SSs, while a Time Division Multiplexing (TDM) mechanism is employed in the downlink subframe. In FDD, the uplink and downlink subframes are concurrent in time, but are transmitted on separate carrier frequencies. Support for half-duplex FDD SSs is also provided, at the expense of some additional complexity. Each subframe is divided into physical slots. Each TDM/TDMA burst carries MAC Protocol Data Units (PDUs) containing data towards SSs or BS, respectively.

The transmission convergence sublayer operates on top of the PHY and provides the necessary interface with the MAC. This layer is specifically responsible for the transformation of variable-lenght MAC PDUs into fixed lenght PHY blocks.[6]

The necessity to provide secure data transmissions has led to the native inclusion of a privacy sub-layer, at the MAC level. Such protocol is responsible for encryption/decryption of the packet payload, according to the rules defined in the standard.[1]

Since IEEE 802.16 uses a wireless medium for communications, the main target of the MAC layer is to manage the resources of the radio interface in an efficient way, while ensuring that the QoS levels negotiated in the connection setup phase are fulfilled. The 802.16 MAC protocol is connection-oriented and is based on a centralized architecture. All traffic, including inherently connectionless traffic, is mapped into a connection which is uniquely identified by a 16-bit address.
The common part sublayer is responsible for the segmentation and the reassembly of MAC service data units (SDUs), the scheduling and the retransmission of MAC PDUs. As such, it provides the basic MAC rules and signalling mechanisms for system access, bandwidth allocation and connection maintenance. The core of the protocol is bandwidth requests/grants management. A SS may request bandwidth, by means of a MAC message, to indicate to the BS that it needs (additional) upstream bandwidth. Bandwidth is always requested on a per-connection basis to allow the BS uplink scheduling algorithm (which is not specified in the standard) to consider QoS-related issues in the bandwidth assignment process.

As depicted in Fig. 1.2, the MAC includes a convergence sublayer which provides three main functionalities:

(1) Classification. The CS associates the traffic coming from upper layer with an appropriate *Service Flow* (SF) and *Connection Identifier* (CID).
(2) Payload Header Suppression (PHS). The CS may provide payload header suppression at the sending entity and reconstruction at the receiving entity.
(3) Delivery of the resulting CS PDU to the MAC Common Part Sublayer in conformity with the negotiated QoS levels.

The standard defines two different Convergence Sublayers for mapping services to and from IEEE 802.16 MAC protocol. The ATM convergence sublayer is defined for ATM traffic, while the packet convergence sublayer is specific for mapping packet-oriented protocol suites, such as IPv4, IPv6, Ethernet and Virtual LAN. As regards IP, the packets are classified and assigned to the MAC layer connections based on a set of matching criteria, including the IP source and the destination addresses, the IP protocol field, the Type-of-Service (TOS) or DiffServ Code Points (DSCP) fields for IPv4, and the Traffic Class field for IPv6. However, these sets of matching criteria are not in the standard and their implementation is left open to vendors.

1.2.1. *QoS Architecture*

As described above, the data packets entering the IEEE 802.16 network are mapped into a connection and a service flow based on a set of matching criteria. These classified data packets are then associated with a particular QoS level, based on the QoS parameters of the service flow they belong to. The QoS may be guaranteed by shaping, policing, and/or proritizing the data packets at both the SS and BS ends. The BS allocates upstream bandwidth for a particular upstream service flow based on the parameters and service specifications of the corresponding service scheduling class negotiated during connection setup. The IEEE 802.16 standard defines four QoS service classes: Unsolicited Grant Service (UGS), Real-Time Polling Service (rtPS), Non-Real Time Polling Service (nrtPS) and Best Effort (BE).[6,7]
These four classes are characterized as follows.

- The UGS service is defined to support constant bit rate (CBR) traffic, such as audio streaming without silence suppression. Unsolicited grants allow SSs to transmit their PDUs without requesting bandwidth for each frame. The BS provides fixed-size data grants at periodic intervals to the UGS flows. Since the bandwidth is allocated without request contention, the UGS provides hard guarantees in terms of both bandwidth and access delay. The QoS parameters defined for this service class are the size of the grant to be allocated, the nominal interval length between successive grants and the tolerated grant jitter, defined as the maximum tolerated variance of packet access delay.
- In the case of Variable Bit Rate (VBR) video traffic, such as MPEG streams, the bandwidth requirements for the UGS grant interval cannot be determined at connection setup time. As a result, peak stream bit rate-based CBR allocation would lead to severe network underutilization, whereas the average bit rate CBR allocation can result in unacceptable packet delay and jitter. The rtPS service has been introduced to accomodate such flows. For this service, indeed, the BS provides periodic transmission opportunities by means of a basic polling mechanism. The SS can exploit these opportunities to ask for bandwidth grants, so that the bandwidth request can be ensured to arrive at the BS within a given guaranteed interval. The QoS parameters relevant to this class of services are the nominal polling interval between successive transmission opportunities and the tolerated poll jitter.

- The non real-time polling services (nrtPS) is similar in nature to rtPS but it differs in that the polling interval is not guaranteed but may depend on the network traffic load. This fits bandwidth-demanding non-real time service flows with a variable packet size, such as large files transfers. In comparison with rtPS, the nrtPS flows has fewer polling opportunities during network congestion, while the rtPS flows are polled at regular intervals, regardless of the network load. In heavy traffic conditions, the BS can not guarantee periodic unicast requests to nrtPS flows, so that the SS would also need to use contention and piggybacking to send requests to the BS uplink scheduler.
- For Best Effort (BE) traffic, no periodic unicast requests are scheduled by the BS. Hence, no guarantees in terms of throughput or packet delay can be given. The BE class has been introduced to provide an efficient resource utilization for low-priority elastic traffic, such as telnet or HTTP.

While these services provide the basics for supporting QoS guarantees, the "real" core, i.e., traffic scheduling, policing, shaping and admission control mechanisms, is not specified by the standard. In the next section, we will present and review some possible QoS architectures for WiMAX-based PMP networks.

1.3. QoS Scheduling in WiMAX Networks

In order to offer an efficient QoS support to the end user, a WiMAX equipment vendor needs to design and implement a set of protocol components that are left open by the standard. These include traffic policing, traffic shaping, connection admission control and packet scheduling.

Due to the highly variable nature of multimedia flows, traffic shaping and traffic policing are required by the SS, in order to ensure an efficient and fair utilization of network resources. At connection setup, the application requests network resources according to its characteristics and to the required level of service guarantees. A traffic shaper is necessary to ensure that the traffic generated actually conforms to the pre-negotiated traffic specification. However, traffic shaping may not guarantee such conformance between the influx traffic and service requirements. This is dealt with by a traffic policer, which compares the conformance of the user data traffic with the QoS attributes of the corresponding service and takes corresponding actions, e.g., it rejects or penalizes non conformance flows.

QoS profiles for SS are usually detailed in terms of Committed Information Rate (CIR) and Maximum Information Rate (MIR) for the various QoS classes.[8,9] The CIR (defined for nrtPS and rtPS traffic) is equal to the information transfer rate that the WiMAX system is committed to carry out under normal conditions. The MIR (defined for nrtPS and BE QoS types) is the maximum information rate that the system will allow for the connection. Both these QoS parameters are averaged over a given interval time.

In order to guarantee that the newly admitted traffic does not result in network overload or service degradation for existing traffic, a (centralized) connection admission control scheme also has to be provided.

Even though all the aforementioned components are necessary in order to provide an efficient level of QoS support, the core of such a task resides in the scheduling algorithm. An efficient scheduling algorithm is the essential *conditio sine qua non* for the provision of QoS guarantees, and it plays an essential role in determining the network performance. Besides, a traffic shaper, policer and connection admission control mechanisms are tightly coupled with the scheduler employed. Therefore, the rest of this section is devoted to such an issue.

Although the scheduling is not specified in the standard, system designers can exploit the existing rich literature about scheduling in wireless ATM,[10] from which WiMAX has inherited many features. If this allows one not to start from scratch, existing schemes need to be adapted to match the peculiar features (e.g., traffic classes, frame structure) of the IEEE 802.16 standard.

As an example, the IEEE 802.16 scheduling mode can be seen as an outcome of the research carried out on hierarchical scheduling.[11] This is rooted in the necessity of limiting the MAC exchange overhead by letting the BS handle all connections of each SS as an aggregated flow. As explained in the previous section, according to the standard, the SSs request bandwidth on per connection-basis; however, the BS grants bandwidth to each individual SS, so that the resources are allocated to the aggregation of active flows at each SS. Each SS is then in charge of allocating the granted bandwidth to the active flows, which can be done in an efficient way since the SS has complete knowledge of its queues status. This, however, requires the introduction of a scheduler at each SS, enhancing the complexity (and consequently the cost) of the SS equipment. A detailed operational scheme is depicted in Fig. 1.3, outlining the role played by each component and the requests/grants mechanism at the basis of WiMAX QoS support.

Schedulers work on multiple connections in order to ensure the negotiated throughputs, delay bounds and loss rates. The target of a scheduling algorithm is to select which connection has to be served next. This selection process is based on the QoS requirements of each connection. An efficient scheduling algorithm at the BS must be provided in order to guarantee proper performance. To better explain the scheduler's role, let us first assume that the BS performs the scheduling functions on a per-connection basis [a]. In order to schedule packets correctly, information such as the number of pending connections, their reserved throughputs and the statues of session queues is needed. While this information is easily accessible as concerns downlink connections, the SSs need to send their bandwidth requests and queue status to the BS for the uplink. This has a twofold effect. On the one hand, it increases the signalling overhead, while, on the other hand, it provides the BS with information that may be not up-to-date (e.g., due to contention delays etc.). In downlink, the scheduler has complete knowledge of the queue status, and, thus,

[a]This was "grant per connection" in the original 2001 standard.

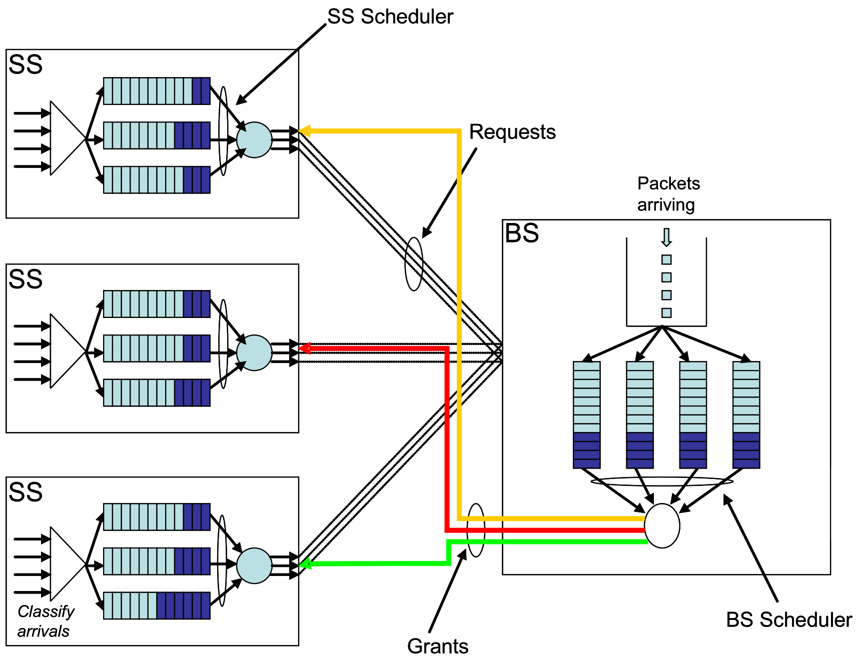


Fig. 1.3. Graphic representation of hierarchical scheduling.

may use some classical scheduling schemes, such as Weighted Round Robin (WRR), Weighted Fair Queueing (WFQ) etc.[10] Priority oriented fairness features are also important in providing differentiated services in WiMAX networks. Through priority, different traffic flows can be treated almost as isolated while sharing the same radio resource. However, due to the nature of WiMAX TDD systems, the BS scheduler is non-work-conserving, since the output link can be idle even if there are packets waiting in some queues. Indeed, after downlink flows are served in their devoted subframe, no additional downlink flows can be served till the end of the subsequent uplink subframe.

Scheduling uplink flows is more complex because the input queues are located in the SSs and are hence separated from the BS. The UL connections work on a request/grant basis. Using bandwidth requests, the uplink packet scheduling may retrieve the status of the queues and the bandwidth parameters. The literature is not rich in terms of QoS scheduling schemes specifically designed for WiMAX networks. In the following, we will briefly describe the most relevant works that address such a topic, to the best of authors knowledge.

In,[12] the authors present a QoS architecture for IEEE 802.16 based on priority scheduling and dynamic bandwidth allocation. In particular, they propose a scheduling process divided into two parts. The first one, executed by the uplink scheduler inside the BS, is performed in order to grant resources to the SSs in response to bandwidth requests. This is done by means of a classical WRR.[13] At each subscriber station, bandwidth assignments are computed by starting from the highest priority class (i.e., UGS flows) and then going down to rtPS, nrtPS and BE. In this way, a strict priority among service classes is guaranteed. The scheduling schemes employed for the various classes are different. A classical WFQ[14] is used for UGS and rtPS, whereas a simpler WRR is used for nrtPS service class. Best Effort traffic is served through a simple FIFO policy. By means of this prioritized approach (which resembles somehow Multiclass Priority Fair Queueing[11]), the proposed architecture is able to guarantee a good performance level to UGS and rtPS classes, to the detriment of lower priority traffic (i.e., nrtPS and BE flows).

Finally, in[7] the authors have assesed, via simulation, the performance of an IEEE 802.16 system using the class of latency-rate[15] scheduling alghoritms where a minimum reserved rate is the basic QoS parameter negotiated by a connection within a scheduling service. Specifically, within this class, they selected defict round robin (DRR) as the downlink scheduler to be implemented in the BS, since it combines the ability to provide fair queuing in the presence of variable lenght packets with the simplicity of implementation. In particular, DRR requires a minimum rate to be reserved for each packet flow being scheduled. Therefore, althought not required by the IEEE 802.16 standard, BE connections should be guaranteed a minimum rate. This fact can be exploited in order to both avoid BE traffic starvation in overloaded scenarios, and let BE traffic take advantage of the excess bandwidth which is not reserved for the other scheduling services. On the other hand, DRR assumes that the size of the head-of-line packet is known at each packet queue; thus, it cannot be used by the BS to schedule transmissions in the uplink direction. In fact, with regard to the uplink direction, the BS is only able to estimate the overall amount of backlog of each connection, but not the size of each backlogged packet. Therefore, the authors selected WRR as the uplink scheduler. Like DRR, WRR belongs to the class of rate-latency scheduling algorithms. At last, DRR is implemented in the SS scheduler, because the SS knows the sizes of the head-of-line packets of its queues.

1.4. A Case Study: Voice over IP Support in WiMAX Networks

In this section we present some preliminary results, obtained from an experimental test-bed, on the ability of WiMAX systems to support Voice over IP (VoIP) applications. The measurements reported below, assess WiMAX capability to support VoIP flows. In particular, the voice quality was evaluated through the E-Model[16] by using the R-factor.[17]

1.4.1. *Testbed Configuration*

Our test-bed is based on Alvarion equipment operating in the (licensed) 3.5 GHz-based frequency band and compliant with the IEEE 802.16d specifications. The experimental data has been collected exploiting a 4-nodes wireless testbed deployed in a rural environment, located in northern Italy, implementing a PMP architecture. The BS is equipped with a sectorial antennas with a gain of 14 dBi covering all the 3 SSs. The default maximum output power at antenna port is 36 dBm for both the BS and the SS. The distance between the BS and SS1, SS2 and SS3 is 8.4 km, 8.5 km and 13.7 km, respectively. The average signal-to-noise ratio is above 30 dB, thus enabling the higher modulation, i.e. 64 QAM, for each connection. The SSs work in line-of-sight conditions under FDD half-duplex. All nodes run a Linux distribution based on a 2.4.31 kernel. The measurements are performed exploiting an Alvarion BreezeMAX platform[18] operating in the 3.5 GHz licensed band and using a 3.5 MHz wide channel in FDD mode. Each node is attached through an Ethernet connection to the WiMAX equipment.

1.4.2. *Parameters Setting*

MIR and CIR are specified for each SS according to the negotiated Service level Agreement (SLA); the compliance to the negotiated SLA is assessed over a reference window, called Committed Time (CT). In what follows we assume that n SSs make MIR and CIR requests to the BS. We let $R_{\max}$ the maximum traffic rate available at the WiMAX Downlink Air Interface, and denote CIR_k and MIR_k the request of the k-th SS[b], where $0 \leq \mathrm{CIR}_k \leq \mathrm{MIR}_k \leq R_{\max}$.
The BS dynamically allocates the BE Service Rate R_{BE} (bit/s) and the Real Time (RT) Service Rate R_{RT} (bit/s) with a cumulative upper bound of $R_{\max}$, making sure that the RT service traffic has a higher priority than the BE service traffic: $R_{\mathrm{RT}} + R_{\mathrm{BE}} \leq R_{\max}$. The residual capacity is allocated as R_{BE}. Let N_{tot} be the total number of downstream service flows consisting of N_{VoIP} VoIP flows and N_{TCP} TCP persistent connection, so that $N_{\mathrm{tot}} = N_{\mathrm{TCP}} + N_{\mathrm{VoIP}}$.
Let $R_{\mathrm{TCP}}(m)$ be the service rate that the BS can provide to the m-th TCP service flow, the aggregated BE service rate is $R_{\mathrm{BE}} = \sum_{m=1}^{N_{\mathrm{TCP}}} R_{\mathrm{TCP}}(m)$; similarly, if $R_{\mathrm{VoIP}}(m)$ is the service rate that the BS provides to the m-th VoIP service flow, the aggregated RT service rate becomes: $R_{\mathrm{RT}} = \sum_{m=1}^{N_{\mathrm{VoIP}}} R_{\mathrm{VoIP}}(m)$. The Alvarion equipment used in the testbed provides resource allocation mechanisms corresponding to three cases.
In the first case, the downlink bandwidth is over-provisioned, meaning that the aggregated traffic service rate for the WiMAX network is *deterministically* lower than $R_{\max}$, i.e. $\sum_{m=1}^{N_{\mathrm{TCP}}} \mathrm{MIR}(m) + \sum_{n=1}^{N_{\mathrm{VoIP}}} \mathrm{MIR}(n) \leq R_{\max}$, and no congestion occurs: the allocation in this case is fairly simple and the BS sets $R_{\mathrm{VoIP}}(n) = \mathrm{MIR}(n)$ and $R_{\mathrm{TCP}}(m) = \mathrm{MIR}(m)$.
The opposite case occurs when the aggregate of the CIR requested by VoIP subscribers exceeds $R_{\max}$, i.e. $\sum_{n=1}^{N_{VoIP}} \mathrm{CIR}(n) > R_{\max}$: then the BS sets $R_{\mathrm{VoIP}}(n) = \frac{R_{\max}}{N_{VoIP}}$ and $R_{\mathrm{TCP}}(m) = 0$ for every SS $n = 1, 2, \ldots, n$.
The remaining case is such that :

[b]The Alvarion BreezeMAX device does not allow to set the MIR parameter for real-time traffic

$$\sum_{m=1}^{N_{\rm TCP}} \mathrm{MIR}(m) + \sum_{n=1}^{N_{\rm VoIP}} \mathrm{MIR}(n) > R_{\max};$$
$$\sum_{n=1}^{N_{\rm VoIP}} \mathrm{CIR}(n) \leq R_{\max}. \tag{1.1}$$

This is the case when the BS guarantees the minimum service rate for the VoIP traffic and can reallocate the remaining bandwidth to the BE services, namely

$$R_{\rm VoIP}(n) = \mathrm{CIR}(n); \tag{1.2}$$
$$R_{\rm TCP}(m) = \frac{(R_{\max} - R_{\rm RT})}{N_{\rm TCP}}.$$

This is also the case that was considered for our measurement, since it is the probing case when QoS guarantees must be provided in spite of concurrent data traffic.

Notice that the actual implementation of the resource allocation depends on the scheduling implemented at the BS and vendors usually do not disclose such a critical detail to customers. Nevertheless, with appropriate probing, we could get some insight into the system behavior (see Sec. 1.4.3). Finally, the IP's DSCP[19] field is exploited in order to enforce a certain QoS class service. Traffic flows belonging to different service categories are tagged using the `iptables` software.[20] During our measurements, all SSs share the same QoS, as summarized in Tab. 1.1.

Table 1.1. Mapping rules of Alvarion BreezeMAX.

Traffic Class	**DSCP**	**CIR (kbps)**	**MIR (kbps)**
BE	1	n.a.	12000
nrtPS	2-31	3000	12000
rtPS	32-63	300	n.a.

1.4.3. *Performance Measurements*

Data flows and VoIP flows were generated via the Distributed Internet Traffic Generator (D-ITG), a freely available software tool.[21] VoIP codecs feed RTP packet flows and two commonly used codecs have been considered, i.e., G.729.2 and G.723.1. VoIP connections are mapped into the rtPS class, while TCP-controlled traffic is mapped into the BE class. Mapping of CBR sources into the rtPS class made much easier to trace the behavior of the system, since the actual scheduling policies were unknown on our side. In order to collect reliable measure of delays, before each experiment we synchronized all nodes using `NTP`.[22] All SSs sustain the same traffic, consisting of a given number of VoIP session plus one persistent TCP connection,

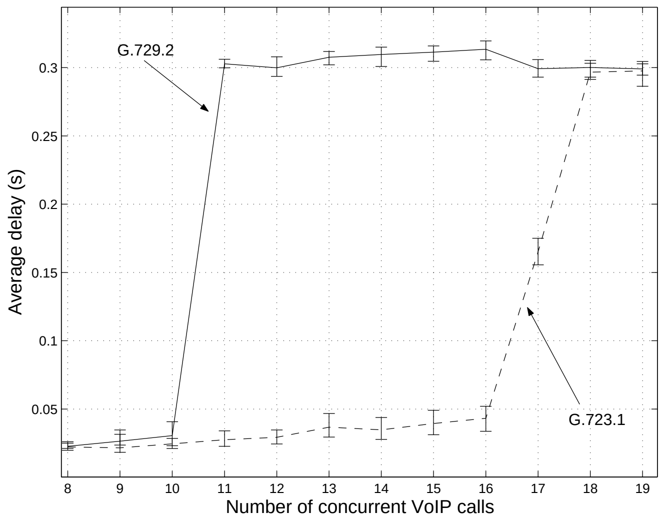


Fig. 1.4. Average delay vs. number of SS VoIP flows; minimum value and maximum value delimiters superimposed.

modeling background traffic. Measurements were performed over 5 minutes intervals and averaged over 10 runs.

In the first set of measurements, we determined the voice capacity, i.e., the maximum number of sustained VoIP calls with high quality ($70 < R < 80$) and related parameters. Here, we report only the downlink results, since it was found to be the bottleneck.

Fig. 1.4 and Fig. 1.5 depict the measurement results we collected for the delay and the packet loss, respectively. Particularly, the delay saturates at 300 ms, whereas, after the saturation point, packet loss increases almost linearly. The G.723.1 codec outperforms clearly G.729.2; such a difference is due to the higher G.729.2 packet generation rate, coupled to the large overhead of packet headers of the RTP/ UDP/IP/MAC protocol stack ($\simeq 45\%$ for the G.729.2). Such effect is well known in VoIP over WLANs: in practice, it is convenient to employ larger speech trunks per packet and consequently larger inter-packet generation intervals.[23,24]

Finally, Fig. 1.6 provides a comprehensive picture in terms of the R-factor. There exist roughly three regions: in the leftmost region, G.729.2 provides a fairly good quality, but after 10 calls, G.723.1 obtains much better performance. In the end, for the given CIR, the system under exam supports up to 17 G.723.1 VoIP calls, and 10 G.729.2 calls per SS.

In order to determine the voice capacity, we restricted to the downlink, claiming that it is the bottleneck. As reported in Fig. 1.7, in fact, the R-factor is higher for the uplink, irrespective of the index of the SS VoIP flow and of the codec.

Furthermore, we sampled the cumulative density function (cdf) of the packet delay around the voice capacity. Fig. 1.8 represents the delay cdf for downlink VoIP flows

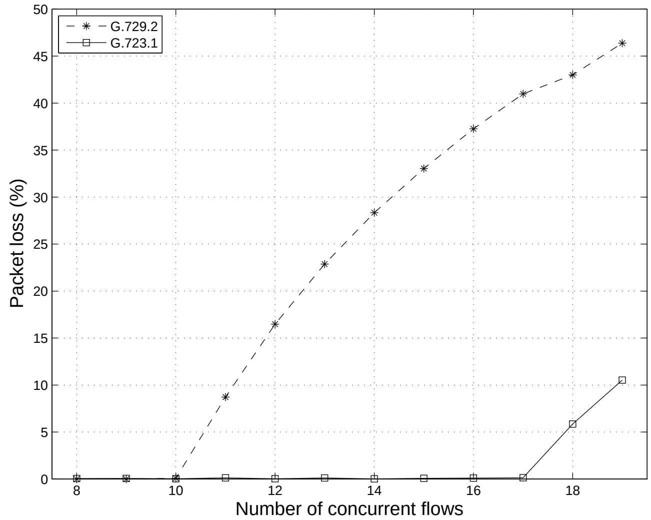


Fig. 1.5. Packet loss rate of VoIP flows per SS using different codecs.

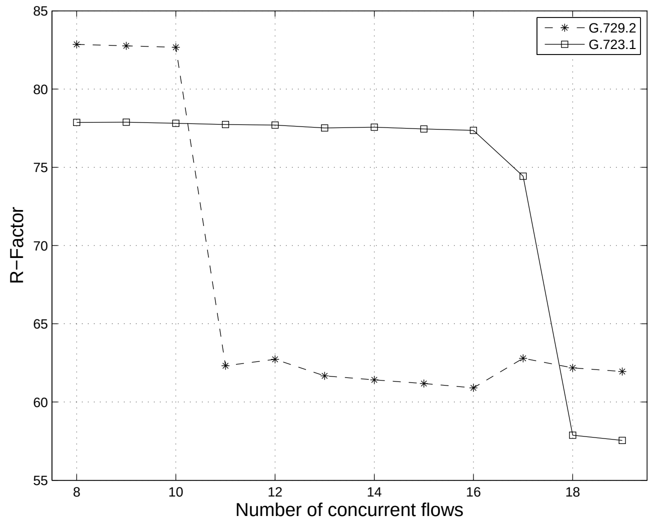


Fig. 1.6. Average R-factor versus number of SS VoIP flows.

using a G.723.1 codec. Even though the scheduling policy is undisclosed, we can infer that it is not simply the average delay to degrade, at the increase of the offered VoIP traffic, but, the whole delay distribution is shifted around higher delay values. The BS operates a strict threshold control policy: in case a SS exceeds a certain threshold above the CIR, all the flows of the violating SS are penalized. Only for 17 G.723.1 VoIP calls the excess above the CIR appears in a delays spreading as

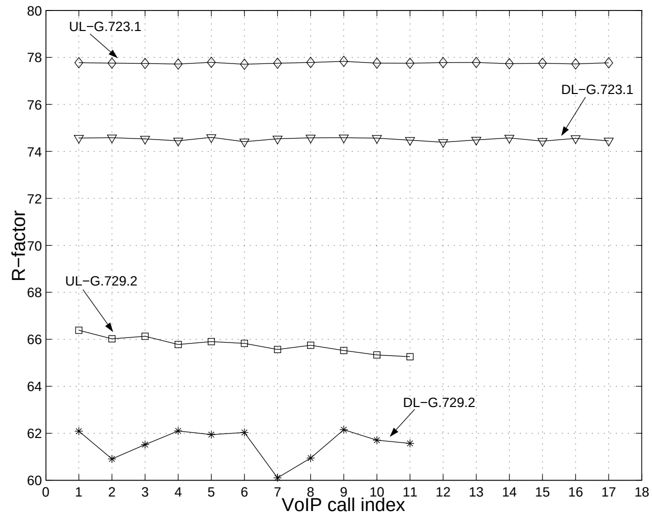


Fig. 1.7. Uplink and Downlink R-factor vs SS VoIP session index, using 11 and 17 concurrent calls with the G.729.2 and G.723.1 codecs, respectively.

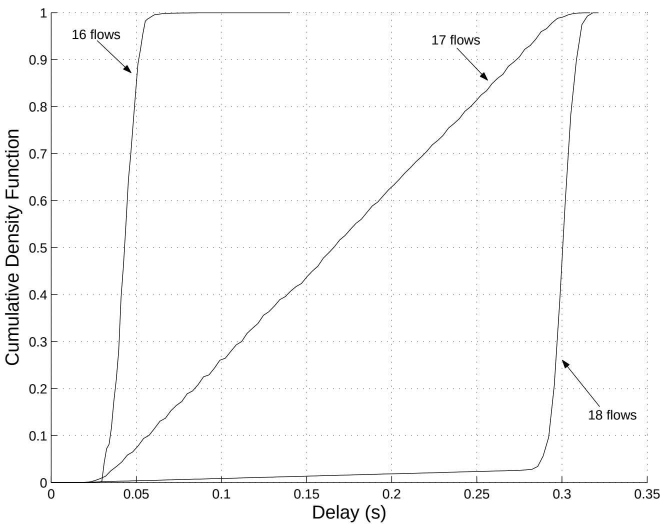


Fig. 1.8. Packet delay cumulative density function, G.723.1 codec, downlink direction.

clearly shown in Fig. 1.8. At the SS side, this strict BS policy calls for admission control of VoIP flows, in order to prevent service degradation. We repeated the same measurements for the uplink and the results were similar. As emerged from the R-factor measurements, the uplink performs better than the downlink. This contradict the simulation results obtained in,[7] where larger uplink delays, compared to the downlink, were ascribed to the bandwidth request mechanism and to the PHY overhead. In the case at hand, the uplink delay due to bandwidth request did not prove significant; we ascribe this fact to the activation of the piggybacking mechanisms for bandwidth reservation provided by WiMAX.

1.5. Research Challenges

Though WiMAX is the most promising technology for enabling BWA systems to be widely deployed, many issues need to be addressed in order to make it effectively support the requirements and constraints of end-users' multimedia flows. In order to do so, according to the discussion mentioned previously, an efficient QoS-enabled scheduling algorithm has to be designed and implemented. In this section, we point out and briefly describe the most promising, as well as challenging, directions in such a field, by outlining a research roadmap for QoS provisioning in WiMAX networks. As we considered the scheduling algorithm in isolation in the last section, we shall now present cross-layer approaches, in which performance improvements are obtained by making an appropriate use of information which comes from the lower and/or upper layers.

- *Multiantenna Architectures for WiMAX Networks.* In recent years, intensive research efforts have led to the development of spectrally efficient multiuser transmission schemes for wireless communications based on the use of multiple antenna systems. The use of multiple antennas in combination with appropriate signal processing and coding is indeed a promising direction which aims to provide a high-data-rate and a high-quality wireless communications in the access link. In this sense, multiantenna systems can be seen as a way to enhance the cell capacity while offering a better and more stable link quality at the same time. On the other hand, antennas arrays can be used also to achieve beamforming capabilities, with a remarkable improvement in terms of network performance. Adaptive Antenna Systems (AAS) are encompassed by the IEEE 802.16 standard to improve the PHY-layer characteristics. However, AAS can also act as enablers of Spatial Division Multiple Access (SDMA) schemes. In this way, multiple SSs, separated in space, can simultaneously trasmit or receive on the same subchannel. This, obviously, demands the realization of a scheduling algorithm able to effectively exploit the presence of such beamforming capabilities. In this way, through a cross-layer approach, striking results can be obtained in terms of QoS support. An AAS-aware scheduling could indeed profit from the additional degree of freedom (i.e., the spatial dimension[25]) provided by the underlying PHY techniques. While this may lead to better performance, it also leads to an increase in the complexity of the scheduler itself. Nonetheless, we believe that the use of this and other related multiantenna techniques (e.g., space-time codes) represent a research direction with big potential in terms of throughput optimization. In order to fully take advantage of the power provided by multiple antenna systems, innovative QoS-enabled scheduling algorithms, able to work in both space and time dimensions, need to be designed and engineered.

- *Opportunistic Scheduling.* In wireless networks, channel conditions may vary over time because of user mobility or propagation phenomena. These effects are usually referred to as shadowing and fading, depending on their typical time-scales. They have been traditionally considered as harmful features of the radio interface due to their potentially negative impact on the quality of communication. However, recent research has shown that the time-varying nature of the radio channel can be used for enhancing the performance of data communications in a multiuser environment. Indeed, time-varying channels in multiuser environments provide a form of diversity, usually referred to as multiuser diversity, that can be exploited by an "opportunistic" scheduler, i.e., a scheduler that selects the next user to be served according to the actual channel status.[26] This approach may also be applied, at the cost of some additional complexity and signalling between PHY and MAC, to WiMAX networks. Opportunistic scheduling schemes do not usually apply to flows that require QoS guarantees, due to the unpredictable delays that may come from the channel dynamics. However, their use may actually lead to an enhanced QoS support. For example, improving the effect of non real-time traffic (i.e., nrtPS and BE traffic) would free some additional resources to higher priority traffic. In this way, opportunistic scheduling schemes may actually help to increase the QoS capabilities of WiMAX networks. Moreover in this case, novel scheduling schemes are required in order to exploit multiuser diversity while providing QoS guarantees to the active traffic flows at the same time. It may be interesting to note that multiple antenna systems can actually be used to build up multiuser diversity by means of random beamforming mechanisms (usually referred to in the literature as "dumb" antennas[27]). While this direction is somehow orthogonal in nature to the one (based on "smart antennas") outlined above, it could be worth investigating whether these two techniques may be implemented to coexist (for example, in a time-sharing fashion) in order to obtain the advantages of both approaches.
- *QoS Support in Mesh-Based Architectures.* The techniques we have presented above as research challenges are aimed at providing a better QoS support in PMP architecture. However, they are still subject to the limits imposed by such an architectural choice in terms of service coverage, network capacity and system scalability. One possible solution to overcome such problems could be the adoption of a mesh-based architecture.[28] In mesh topologies, direct communication among neighboring SSs is allowed, so enhancing the network coverage and possibly enabling the deployment of a fully wireless backbone connecting to an Internet gateway. While mesh-based architectures offer interesting possibilities thanks to its inherent flexibility, they also present many research challenges

to be addressed in terms of medium access control and packet routing. This is even more challenging in the case of QoS support for multimedia flows, where reliable levels of services have to be ensured by means of distributed algorithms. In this framework, a "double cross-layer" approach (where information is shared among PHY, MAC and NET layers) may lead to potentially dramatic performance improvements compared to conventional layered solutions. This clearly entails the definition of radically innovative scheduling protocols, which are able to work in a distributed and collaborative way, so cooperating with the routing algorithms in order to provide QoS guarantees to service flows based on some PHY information. For example, the integration of scheduling and routing protocols can be based on the actual channel conditions, as well as on the level of interference in the network [c]. The application of these concepts to WiMAX networks is not straightforward, since it would imply some major modifications to the actual standard, in terms of both signalling (necessary for pursuing cross-layer optimization) as well as definition of basic functionalities and interfaces of the routing protocol to be employed.

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[c]Note that this requires the introduction of novel metrics for path selection in routing algorithms.[28]

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