Publication P6

Jani Lakkakorpi and Alexander Sayenko. 2008. Backhaul as a bottleneck in IEEE 802.16e networks. In: Proceedings of the 2008 IEEE Global Telecommunications Conference (GLOBECOM 2008). New Orleans, Louisiana, USA. 30 November - 4 December 2008, pages 1-6.

Backhaul as a Bottleneck in IEEE 802.16e Networks

Jani Lakkakorpi
Nokia Devices R&D
Helsinki, Finland
jani.lakkakorpi@iki.fi

Alexander Sayenko
Nokia Research Center
Helsinki, Finland
alexander.sayenko@nokia.com

Abstract—**In this paper, we assume that the bottleneck in IEEE 802.16e networks is not necessarily the air interface but it can sometimes be the backhaul, too, since overprovisioning is not always feasible. Our simulations show that applying Differentiated Services and Active Queue Management on the bottleneck links as well as taking the bottleneck link load into account in connection admission control decisions leads into more efficient use of the scarce backhaul capacity.**

Keywords-active queue management; admission control; differentiated services; IEEE 802.16; quality of service; WiMAX

I. INTRODUCTION

WiMAX (Worldwide Interoperability for Microwave Access) is a telecommunications technology aimed at providing wireless data over long distances in a variety of ways, from point-to-point links to full mobile cellular type of access. WiMAX is based on IEEE 802.16 standard; IEEE 802.16-2004 [1] was introduced first and it was followed by the 802.16e-2005 amendment [2]. The former is generally referred to as “Fixed WiMAX” while the latter is known as “Mobile WiMAX”. Terms 802.16d and 802.16e are being widely used, too. IEEE 802.16d uses OFDM PHY as the air interface technology while IEEE 802.16e uses OFDMA PHY. In this paper, we focus on IEEE 802.16e only.

The 802.16e standard supports mesh and point-to-multipoint (PMP) operation modes. In the mesh mode, subscriber stations (SS) communicate to each other and to the base stations (BS) whereas in the PMP mode, the SSs communicate through the BS. It is very likely that the network operators will use the PMP mode to connect their customers to the Internet. Thus, the network operator can control the environment to ensure the Quality of Service (QoS) requirements of its customers.

At the IEEE 802.16e BS, all downlink (DL) connections have dedicated buffers and resources are allocated per connection. There can be multiple connections between the BS and an SS. In uplink (UL) direction, however, the BS grants slots per SS (GPSS) and not per connection (GPC). The SS then decides how the resources are shared between different UL connections.

IEEE 802.16e has five scheduling service classes. In unsolicited grant service (UGS), the BS allocates fixed-size grants periodically; UGS connections do not send any bandwidth requests. In real-time polling service (rtPS), the BS periodically polls the SS by granting one slot for sending a bandwidth request, while the goal of extended real-time polling service (ertPS) is to combine the advantages of UGS and rtPS. In ertPS, the BS continues granting the same amount of bandwidth (by default, the size of this allocation corresponds to maximum sustained traffic rate of the connection) until the ertPS connection explicitly requests a change in polling size. Extended piggyback request field of the grant management subheader can be used for this purpose. If the request size is zero, the BS may provide allocations for bandwidth request header only or nothing at all. In the latter case, contention request opportunities may be used. Non-real time polling service (nrtPS) is similar to rtPS except that connections are polled less frequently and they can use contention request opportunities. Best Effort (BE) connections are never polled but they can receive resources only through contention.

For each scheduling service class there is a corresponding data delivery service class (see Table I). The service classes are defined for UL direction only whereas the data delivery service classes are defined for both UL and DL direction. QoS parameters are almost identical for a scheduling service and a corresponding data delivery service.

TABLE I. SCHEDULING SERVICES AND CORRESPONDING DATA DELIVERY SERVICES

Scheduling service	Corresponding data delivery service
UGS	Unsolicited Grant Service (UGS)
rtPS	Real-Time Variable-Rate Service (RT-VR)
ertPS	Extended Real-Time Variable Rate Service (ERT-VR)
nrtPS	Non-Real Time Variable Rate Service (NRT-VR)
BE	Best Effort Service (BE)

This paper studies the backhaul as a bottleneck in IEEE 802.16e networks. One likely reason for a possible bottleneck in the backhaul is femtocells, which are currently discussed actively in the research community and in the WiMAX Forum [3]. The general idea is that a low cost BS is connected to the customer wired network, thus providing a local access point similar to Wi-Fi hotspots. Then, it is possible that the wired connection can be of lower bandwidth when compared to the maximum bandwidth achieved at the wireless interface.

In our studies, we run different simulation scenarios, where we apply Differentiated Services (DiffServ) and Active Queue Management (AQM) on the bottleneck links. Bottleneck link load based connection admission control (CAC) is studied, too.

The rest of this paper is organized as follows: section 2 presents the proposed techniques, sections 3 and 4 present our

This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE "GLOBECOM" 2008 proceedings.

simulator and the simulation results, respectively, while section 5 concludes the paper.

II. Backhaul Provisioning, Differentiated Services and Connection Admission Control

The simplest and most straightforward approach to backhaul QoS is overprovisioning. Naturally, this is not always feasible, since the bottlenecks could be microwave radio links or leased lines and thus the cost of overprovisioning would be prohibitive.

A. *Differentiated Services*

DiffServ [4] can alleviate the problem: we can prioritize real-time traffic over non-real time traffic. Expedited Forwarding (EF) [5] can be used for the real-time traffic, while Assured Forwarding (AF) [6] or Best Effort (BE) can be used for the non-real time traffic. In practice, EF per-hop behavior (PHB) is usually implemented as a priority queue with a rate limiter, while AF and BE queues have scheduling weights. As long as the real-time traffic load is low enough (e.g., less than 50% of the bottleneck link capacity), there is no need for immediate bottleneck link upgrades. However, admission control based on bottleneck link load is needed – otherwise all real-time connections will suffer during periods of congestion.

Mapping from IEEE 802.16e data delivery services to DiffServ traffic classes should be kept as simple as possible (simple guidelines on mapping services to DiffServ classes have been presented in [7]). No gains can be achieved from having more than three DiffServ classes if the BS scheduler has only three priority levels. Thus, we suggest the following mapping:

- UGS, ERT-VR and RT-VR are mapped to EF
- NRT-VR is mapped to AFx[1] or Best Effort
- BE is mapped to Best Effort

Having two DiffServ classes only is a solution that requires no scheduling weight tuning. If we have three DiffServ classes, we need to set appropriate scheduling weights for AFx and BE. Strict priority like weights for AFx and BE, e.g., 90:10, could be a good starting point.

Most routers that support DiffServ have also support for AQM. Usually, this means applying Random Early Dropping (RED) [8] for the AF and BE queues. In the AF queues, we naturally need one set of thresholds per drop precedence level. If configured properly, RED keeps the router queues short (without degrading TCP goodput) and thus reduces the end-to-end delay. AQM can also be applied at the WiMAX BS DL queues, where it can be done per connection. We have shown good results (considerable reduction in delay) on this in [9].

B. *Connection Admission Control*

As in the case of other wireless technologies, it is likely that the radio interface will be the biggest system bottleneck with WiMAX, too. Thus, the amount of real-time connections has to be controlled in order to guarantee the QoS. This can be done at the BS, for example, by monitoring the DL queuing delays, virtual UL queue sizes and the number of free slots for real-time traffic. However, it seems that only the number of free slots is a reasonable choice for UL admission control as the virtual (mostly bandwidth request based) UL queue sizes may not always be that accurate.

In each frame, when all real-time connections have been served, we check the number of remaining DL and UL slots for real-time connections (our scheduler may save some slots for BE traffic only). Then, we update the averages, which are used in admission control and compare them to our free slots threshold (e.g., 10 slots). w_S is a configurable averaging weight; it determines how fast the averaged number of free slots changes over time.

$$freeSlotsAv_t = (1 - w_S) * freeSlotsAv_{t-1} + w_S * freeSlots_t \quad (1)$$

Since using the averaged number of free slots as such offers no protection against connections arriving in large batches, we choose a more conservative approach instead and exercise bookkeeping with dynamically updated reservation limits. A similar method has been presented in [10].

Whenever a real-time connection arrives, we check if the sum of currently used bandwidth and the MRTR (Minimum Reserved Traffic Rate) of the connection is below the corresponding limit. If this is the case, the connection is admitted and the MRTR is added to the used DL/UL bandwidth. The reservation limits are updated periodically and they are based on the averaged number of free slots for real-time traffic. Fig. 1 illustrates the updating algorithm.

```
if (freeSlotsAv > highTh)&&(limit < maxBw)
    limit := limit + increment

if (freeSlotsAv < lowTh)
    limit := limit * coefficient
    if (limit < reservedBw)
           limit := reservedBw
```

Figure 1. Reservation limit updating algorithm for DL and UL.

If backhaul can be a bottleneck, we should definitely take it into account in connection admission control decisions. This can be done, for example, by monitoring the bottleneck link real-time traffic loads, i.e., the number of delivered bits per second – averaged in a similar way as in (1). The mechanism can be more detailed, too, as presented in [10].

III. WiMAX Network Simulation Model

The basic implementation of our WiMAX module is described in [11]. The module includes the following features: OFDM and OFDMa PHY levels, hybrid ARQ (HARQ), transport and management connections, fragmentation (a large packet can be split between several MAC PDUs), packing (several packets and packet fragments can be put into a single PDU), ranging and bandwidth request contention periods, CDMA codes for ranging and bandwidth requests, support for the most important MAC level signaling messages and the ARQ mechanism that allows retransmitting dropped PDUs.

[1] There are four AF classes: x is an integer between 1 and 4.

Additionally, the module includes several different BS schedulers and it has a simple model for link adaptation. These features are described in more detail in the following sections.

A. MCS, Link Adaptation and Errors

Modulation and coding scheme (MCS) defines how many bits can be sent in a single slot. The BS can dynamically change both the DL and UL MCS of an SS. Link adaptation is based on reported SNR values and carefully tuned transition thresholds. Naturally, we have a different set of link adaptation thresholds for HARQ and non-HARQ connections. Our error model analyzes the PDU SNR, maps it to the FEC BLER based on the channel performance curves, and decides whether the PDU is erroneous or not. Each SS has its own, randomly selected, trace file, where the SNR values are read[2]. We have obtained our SNR values from system simulations.

B. Scheduler

The BS scheduler grants slots for the SSs according to the QoS parameters and bandwidth request sizes of the individual connections. Uplink virtual queue sizes are updated based on bandwidth requests (that sometimes include the real SS queue size) and received UL packet sizes. For DL connections, we use the BS queue sizes and the QoS parameters. In our basic scheduler, slots are assigned in DRR [12] fashion. Quantum size is a configuration parameter (default is 17 slots for all connections); a bigger quantum size decreases the MAP overhead as we then serve fewer connections per frame.

We have implemented support for three IEEE 802.16e data delivery service classes: ERT-VR, RT-VR and BE. ERT-VR and RT-VR connections are served first; they are assigned slots until all ERT-VR and RT-VR queues are empty or until there are no more slots left for real-time traffic. In order to avoid the starvation of BE connections, we can reserve a portion of all slots exclusively for these connections. Moreover, connection admission control should take care that there are always enough slots for real-time connections.

rtPS connections are polled regularly (frequency depends on connection parameters), because they cannot participate in contention like the UL BE connections. ertPS connections are polled regularly, too, and during active periods they are regularly granted slots according to their QoS requirements.

In order to waste as little bandwidth as possible, silence suppression detection at the BS is done for the ertPS connections: whenever an UL frame is received, a connection-specific timer is launched. When this timer expires, we go into silence state, where only polling slots are granted. We do not let the ertPS connections participate in contention, because that might introduce large medium access delays.

BE connections are served after ERT-VR and RT-VR connections; they are served until the BE queues are empty or until there are no slots left. If there are still available slots, they are assigned for contention.

If ARQ is enabled for a connection, the following connection-internal scheduling order is applied (at the SS, too): 1) ARQ feedback messages, 2) retransmissions and 3) all other PDUs. We simulate only cases with one UL/DL transport connection per SS.

[2] 60% of our trace files correspond to ITU PedB model and 40% to ITU VehA model.

IV. Performance Evaluation

We use a modified version of the ns-2 simulator [13]. The WiMAX related modifications have been described in the preceding sections. Six simulations are run in each case in order to obtain 95% confidence intervals. Simulation time is always 200 seconds. One-way core network delay between a server and the BS is set to 31 ms (see Fig. 2). The most important WiMAX network parameters are listed in Table II.

TABLE II. WiMAX Network Parameters

Parameter	Value
PHY	OFDMa
Bandwidth	10 MHz
FFT size	1024
Cyclic prefix length	1/8
TTG (Transmit-receive Transition Gap)	296 PS
RTG (Receive-transmit Transition Gap)	168 PS
Duplexing mode	TDD
Frame length	5 ms
DL/UL ratio	35/12
DL/UL permutation zone	FUSC/PUSC
Channel report type and interval	CQICH, 20 ms
MAP MCS	QPSK-1/2
Compressed MAP	Yes
Number of ranging opportunities	1
Ranging backoff start/end	0/15
Number of request opportunities	3
Request backoff start/end	3/15
CDMA codes for ranging and bw requests	64/192
HARQ	CC, for VoIP connections only
Number of HARQ channels	16
HARQ buffer size	2048 B per channel
HARQ shared buffer	Yes
Max. number of HARQ retransmissions	4
HARQ ACK delay	1 frame
PDU SN	HARQ: ON, HARQ/ARQ: OFF
Fragmentation/Packing	Yes/Yes
Maximum MAC PDU size	100 bytes
ARQ	For all but VoIP connections
ARQ feedback types	All
ARQ block size / window size	64 bytes / 1024
ARQ block rearrangement	No
ARQ feedback frequency	5 ms
ARQ retry timer	50 ms
ARQ block lifetime	1500 ms
ARQ rx purge timeout	2000 ms

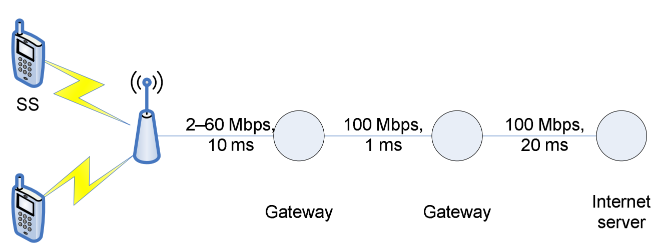

Figure 2. Simulation topology.

In all but admission control simulations, we simulate the following traffic mix: 5 VoIP connections, 5 video streaming connections (DL only); 10, 14, 18, 22, 26 or 30 web browsing connections and 5, 7, 9, 11, 13 or 15 file downloading connections per BS. All user traffic is given BE treatment except for VoIP traffic that is given RT-VR treatment.

Our G.723.1 VoIP traffic source is a simple On-Off Markov model, where both On and Off period lengths are exponentially distributed with mean lengths of 1.2 s and 0.8 s, correspondingly. 24 bytes of payload is sent every 30 ms during active periods. Altogether, RTP, UDP and IP add 40 bytes of overhead, which results in a total packet size of 64 bytes. Simple header compression (from 40 bytes to 4 bytes) is applied both at the BS and at the SS.

The variable bit rate video traffic source is simulated according to [14]. The following parameters are used: mean rate of 125 kbps, maximum rate of 250 kbps and 1500-byte packets. UDP is used as a transport protocol and IPv4 adds 20 bytes of overhead, which results in a payload size of 1472 bytes.

Our web browsing traffic source is simplified; it is necessary to limit the variability of web page sizes as TCP goodput depends on that, too. We have a main page and 30 inline items per page, all items have a size of 4.91 kB, which results in a total page size of 152 kB (this is based on our own measurements). Page reading time is uniformly distributed between 1 and 5 seconds and four NewReno TCP connections are utilized. The same HTTP traffic model is used for modeling file downloading, but in that case only a single 250 kB file is downloaded. Time between two downloads is uniformly distributed between 1 and 5 seconds. A single NewReno TCP connection is utilized.

In our admission control simulations, we simulate the following traffic mix: a variable number of VoIP connections and 10 file downloading connections per BS. New VoIP connections arrive to the system, according to a Poisson process, with an intensity of 6.7 connections per second. However, 200 first connections arrive with an intensity of 20 connections per second and without admission control. VoIP connection duration is exponentially distributed with a mean of 60 seconds, while the file downloading connections are active during the whole simulation run.

Depending on the simulated case, a new VoIP connection is admitted to the network only if:

1. Target number of free DL/UL real-time slots is 20, 15, 10 or 5. AC parameters: $w_S = 0.001$, *highTh* = 22, 12, 7 or 5, *lowTh* = 18, 8, 3 or 1, *increment* = 3000, *coefficient* = 0.9, *maxBw* = 3 Mbps, *MRTR* = 7467 bps, limit update frequency = 100 ms.
2. The same conditions as above and real-time bottleneck link load is less than 0.45 both in DL and UL direction (sample period: 100 ms, averaging weight: 0.1).

Fig. 3–11 illustrate the main results of our DiffServ and AQM simulations. In Fig. 3, BE VoIP delay grows as the link speed decreases, while in Fig. 4 EF VoIP delay stays the same as the link speed decreases. Fig. 5 illustrates an interesting phenomenon: when the number of users increases, the backhaul is not so big a bottleneck anymore; there is now more overhead (and VoIP header compression) on the air interface and this overhead does not load the backhaul. Naturally, the same phenomenon would happen if VoIP packets were given BE treatment on the backhaul (this can actually be seen in Fig. 3, where the BE VoIP delay with 10 Mbps bottleneck starts to decrease when the number of users grows big enough).

Fig. 6 and Fig. 7 show that AQM does not really affect TCP goodput. However, in Fig. 9 and Fig. 10 we can see an improvement in DL delay when AQM is applied at the bottleneck link router output queues and/or at the BS per-connection queues. UL delay performance is also slightly better due to fewer TCP retransmissions, which in turn results in fewer TCP acknowledgments.

Fig. 8 and Fig. 11–13 show that when the bottleneck link load is taken into account in real-time connection admission control decisions, there is no need for overprovisioning. When the real-time traffic load on the bottleneck link exceeds its admission control threshold, no new real-time connections are admitted to the system (naturally, if this happens frequently, it is time to upgrade the bottleneck link). Fig. 8 shows that when the radio resource based admission control is less strict (target number of free UL slots for real-time traffic is 15 or less), backhaul becomes the actual bottleneck. It can be seen in Fig. 11 that this results in higher real-time traffic packet loss (excess EF packet are dropped in the rate limiters) and thus larger number of unsatisfied users.

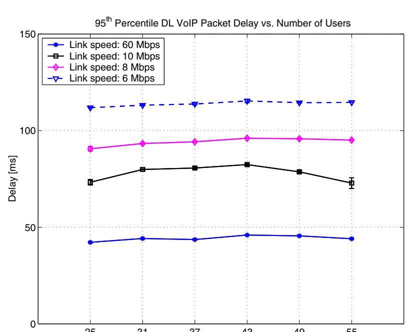

Figure 3. BE VoIP: 95th percentile DL VoIP delay.

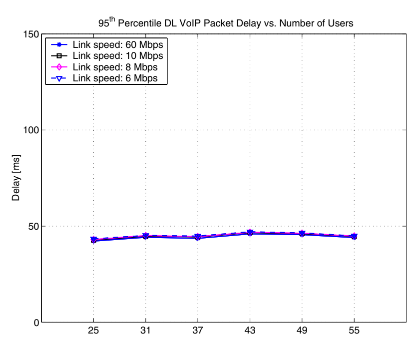

Figure 4. EF VoIP: 95th percentile DL VoIP delay.

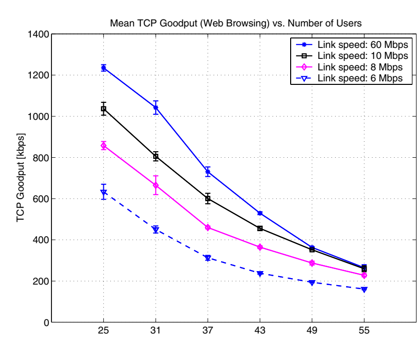

Figure 5. EF VoIP: TCP goodput for web browsing.

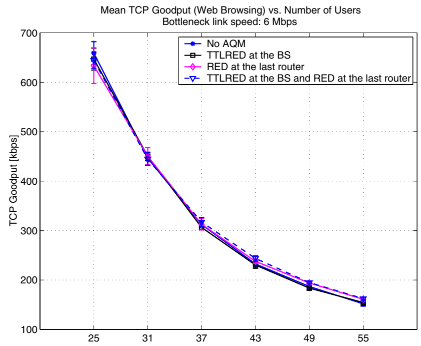

Figure 6. 6 Mbps bottleneck: TCP goodput for web browsing.

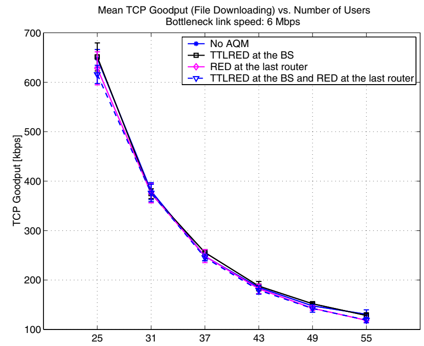

Figure 7. 6 Mbps bottleneck: TCP goodput for file downloading.

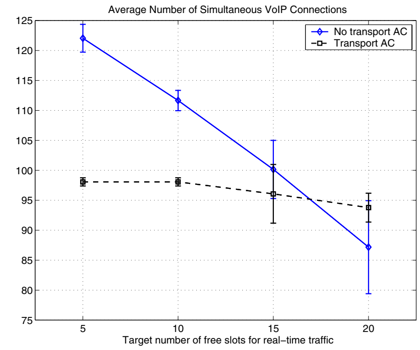

Figure 8. 2 Mbps bottleneck: number of users.

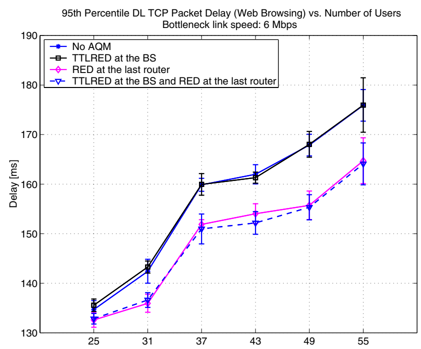

Figure 9. 6 Mbps bottleneck: 95th percentile DL delay for web browsing.

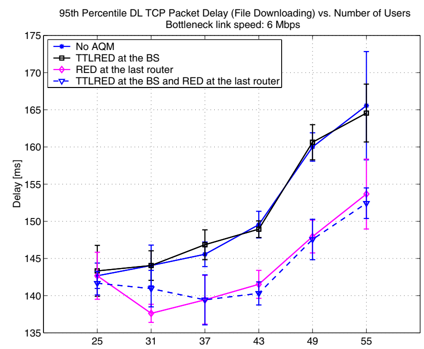

Figure 10. 6 Mbps bottleneck: 95th percentile DL delay for file downloading.

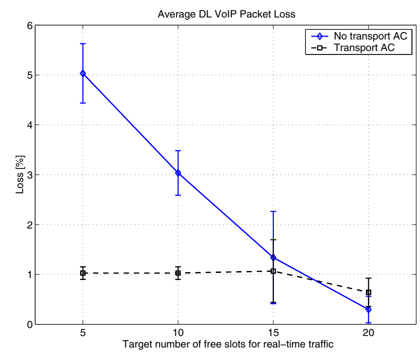

Figure 11. 2 Mbps bottleneck: DL VoIP packet loss.

Fig. 12 and Fig. 13 illustrate the real-time traffic load in DL and UL direction on the bottleneck link as well as the number of real-time connections and the number of free slots for UL real-time traffic. This is the case, where the target number of free real-time slots is five and thus backhaul is the bottleneck. The real-time traffic load on the bottleneck link cannot exceed 50% of the link bandwidth due to EF rate limiter that drops the excess traffic.

V. Conclusions

In this paper, we have shown that there is no need for major overprovisioning of the backhaul but we can (and should) apply well-known IP traffic management methods in order to control the bottleneck link loads[3].

DiffServ can help: If VoIP and other real-time traffic is given priority over non-real time traffic, there is no need for immediate bottleneck link upgrades. Also, real-time traffic load of the bottleneck link should be taken into account in VoIP admission control decisions.

We think that mapping from WiMAX data delivery service classes to DiffServ traffic classes should be kept as simple as possible; no gains can be achieved from having more than three DiffServ classes if the BS scheduler has only three priority levels. Moreover, network management becomes increasingly difficult when there are more classes.

AQM can be applied both for the connection-specific DL buffers at the BS and for the traffic aggregates in backhaul routers. AQM reduces delay considerably, especially when applied at the BS.

Acknowledgment

The authors would like to thank everyone involved in the ns-2 simulator development at the Telecommunication laboratory of University of Jyväskylä.

[3] Naturally, things may be different if the backhaul carries some background IP traffic, too.

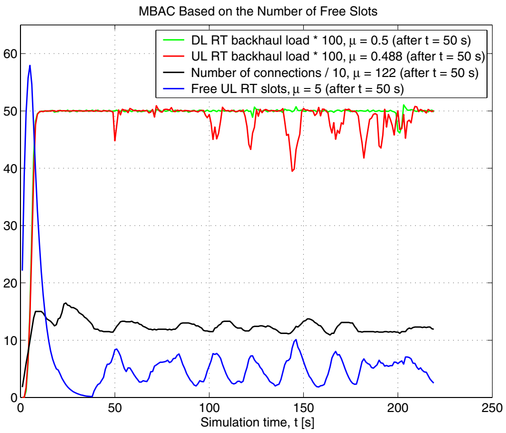

Figure 12. 2 Mbps bottleneck: admission control based on radio load only.

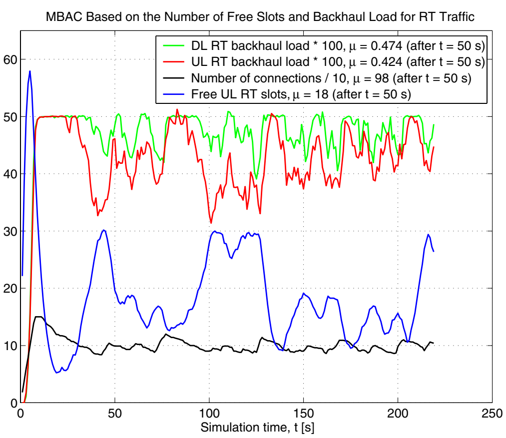

Figure 13. 2 Mbps bottleneck: admission control based on radio and bottleneck link load.

REFERENCES

[1] Air interface for fixed broadband wireless access systems. IEEE Standard 802.16, Jun. 2004.

[2] Air interface for fixed broadband wireless access systems – amendment for physical and medium access control layers for combined fixed and mobile operation in licensed bands. IEEE Standard 802.16e, Dec. 2005.

[3] WiMAX Forum, "Requirements for WiMAX Femtocell Systems," version 0.3.0, Jul. 2008.

[4] S. Blake, D. Black, M. Carlson, E. Davies, Z. Wang and W. Weiss, "An Architecture for Differentiated Services," RFC 2475, Dec. 1998.

[5] B. Davie, A. Charny, J.C.R. Bennett, K. Benson, J.Y. Le Boudec, W. Courtney, S. Davari, V. Firoiu and D. Stiliadis, "An Expedited Forwarding PHB," RFC 3246, Mar. 2002.

[6] J. Heinänen, F. Baker, W. Weiss and J. Wroclawski, "Assured Forwarding PHB Group," RFC 2597, Jun. 1999.

[7] J. Babiarz, K. Chan and F. Baker, "Configuration Guidelines for DiffServ Service Classes," RFC 4594, Aug. 2006.

[8] S. Floyd and V. Jacobson, "Random Early Detection Gateways for Congestion Avoidance," IEEE/ACM Transactions on Networking, vol. 1, pp. 397–413, 1993.

[9] J. Lakkakorpi, A. Sayenko, J. Karhula, O. Alanen and J. Moilanen, "Active Queue Management for Reducing Downlink Delays in WiMAX," Proceedings of IEEE VTC2007-Fall, Baltimore, MD, USA, Oct. 2007.

[10] J. Lakkakorpi, O. Strandberg and J. Salonen, "Adaptive Connection Admission Control for Differentiated Services Access Networks," IEEE Journal on Selected Areas in Communications, vol. 23, pp. 1963–1972, Oct. 2005.

[11] A. Sayenko, O. Alanen, J. Karhula and T. Hämäläinen, "Ensuring the QoS Requirements in 802.16 Scheduling," Proceedings of IEEE/ACM MSWiM 2006, Torremolinos, Spain, Oct. 2006.

[12] M. Shreedhar and G. Varghese, "Efficient fair queueing using Deficit Round-Robin," IEEE/ACM Transactions on Networking, vol. 4, pp. 375–385, Jun. 1996.

[13] UCB/LBNL/VINT, "Network Simulator – ns (version 2)," Mar. 2008.

[14] B. Maglaris, D. Anastassiou, P. Sen, G. Karlsson and J. Robbins, "Performance Models of Statistical Multiplexing in Packet Video Communications," IEEE Transactions on Communications, vol. 36, pp. 834-844, Jul. 1988.