Increasing the VoIP Capacity through MAP Overhead Reduction in the IEEE 802.16 OFDMa Systems

Vitaliy Tykhomyrov

Telecommunication laboratory, University of Jyväskylä, Finland
vitaliy.tykhomyrov@jyu.fi

Abstract. One of the main issues with supporting VoIP service over 802.16 networks is the signalling overhead caused by the downlink MAP messages due to frequent transmissions and small packets. To decrease the MAP overhead, the 802.16 standard proposes some mechanisms, such as the compressed MAP and sub-MAPs. In this paper, we show by means of extensive dynamic simulations that sub-MAPs can reduce dramatically the signalling overhead associated with VoIP traffic and significantly improve overall VoIP capacity. At the same time, since sub-MAPs are more sensitive to packet drops, they tend to increase the number of HARQ retransmissions in downlink and transmission delays in the uplink direction.

Keywords: IEEE 802.16, VoIP, Signalling overhead.

1 Introduction

IEEE 802.16 is a standard for wireless broadband access networks [1] that can provide a high-speed wireless access to the Internet for home and business subscribers. It provides the missing link for the *last mile* connection in metropolitan area networks where DSL (Digital Subscriber Line), Cable and other broadband access methods are not available or are too expensive. IEEE 802.16 also offers an alternative to satellite Internet services for rural areas and allows mobility of the customer equipment. The standard is concerned with the air interface between a subscriber station (SS) and a base station (BS).

IEEE 802.16 has been developed before other beyond-3G standards, such as LTE (Long Term Evolution), but there are a few key areas where the performance of 802.16 systems can be improved. Voice over IP (VoIP) service is one of them. One of the key areas for the current revision of 802.16 is the increasing VoIP capacity. The VoIP capacity of any 802.16 OFDMa system is related to the associated overhead. Overhead is important for VoIP applications due to the frequent transmission and small packet size. In the 802.16 system, much of

P. Chatzimisios et al. (Eds.): MobiLight 2010, LNICST 45, pp. 590–603, 2010.

the overhead associated with VoIP traffic occurs in the MAP messages, since dynamic scheduling is used to support VoIP.

To decrease the MAP overhead, the 802.16 standard [2] proposes a few mechanisms, such as the compressed MAP and sub-MAPs. Further enhancements to the 802.16 system [5] introduce persistent allocations.

In this paper we study the sub-MAPs in VoIP applications and analyze how the sub-MAPs can reduce the overhead associated with VoIP traffic and improve overall VoIP capacity.

MAP overhead reduction in VoIP applications has not been studied thoroughly, especially in the IEEE 802.16 networks. In [7], the authors propose a new optimization model and evaluate the performance of a dynamic OFDMa system with inband signalling in terms of how the signalling overhead affects the system performance. However, they consider the signalling overhead only in the downlink direction, even though the signalling overhead of both the downlink and uplink affects the system throughput. In [12], the authors address the problem of allocation representation of VoIP packets and propose an efficient uplink mapping scheme that decreases the size of the allocation map in the IEEE 802.16e OFDMa system. They also present analytical and simulation models to evaluate the performance of the VoIP services. Their results show that the signalling overhead is greatly decreased when the proposed mapping scheme is used. However, the proposed mapping scheme does not conform to the IEEE 802.16 standard since it requires some modifications. Furthermore, their simulator implementation does not contain all the main features of the IEEE 802.16 standard, such as error modeling. In [6], the authors focus on features and solutions used in the IEEE 802.16 standard to support voice traffic and optimization concepts such as persistent allocation. In order to study VoIP performance, they implemented the persistent scheduling scheme in a system-level simulator. The authors state that their simulator complies with the baseline configuration and simulation assumptions in the 802.16m evaluation methodology [3], but they do not provide any description of what is implemented. In [14], the authors consider an AMC scheme for the sub-MAPs to reduce the MAP signaling overhead without explicit information on the channel condition. Indeed, the proposed scheme achieves the same coverage as the broadcast MAP while significantly enhancing the VoIP capacity. However, the authors do not describe their implemented simulation environment, and therefore there is no information whether it does contain all the main features of the IEEE 802.16 standard. Moreover, the authors do not consider the IEEE 802.16m Evaluation Methodology Document [3]. In order to evaluate the MAP overhead, the authors use only 4 MCS levels, while the basic 802.16e system operates 11 different MCS levels.

The rest of this paper is organized as follows. Section 2 provides an insight on VoIP scheduling in the IEEE 802.16. Section 3 presents several simulation scenarios and simulation results. Finally, Section 4 concludes the article and outlines further research directions.

2 VoIP Scheduling and Sub-MAPs in IEEE 802.16

2.1 802.16 Scheduler Implementation

Extended real-time polling service (ertPS) is designed to support real-time service flows that generate variable-size data packets on a periodic basis, such as Voice over IP services with silence suppression. On the downlink, the BS is smart enough to control directly the scheduling of VoIP traffic and allocation of the resources. For the uplink, the BS periodically assigns UL resources according to the requested size, until the SS requests another bandwidth size.

To obtain dynamic simulation results we have implemented our own 802.16 BS scheduler in the NS-2 simulator [11]. The 802.16 BS scheduler accounts for connection parameters, scheduling class type, modulation and coding scheme (MCS), the queue size, and the bandwidth request size. Depending on the direction, the scheduler analyzes either the bandwidth request size sent by the SS or the queue size at the BS.

As for the ertPS class, the 802.16 scheduler allocates N slots for each VoIP connection:

$$N = \begin{cases} 0, & \text{if } R_{size} = 0 \\ \frac{B*G}{S*FPS}, & \text{if } R_{size} > 0, \end{cases} \tag{1}$$

where R_{size} is a bandwidth request size sent by the SS (or the downlink queue size), B is a bandwidth requirement which corresponds to the maximum sustained traffic rate of the VoIP connection QoS profile, S is a slot size as governed by the connection MCS, which determines the number of bytes the SS can send within one slot, FPS represents the number of 802.16 frames per second, and finally G is a grant interval in frames. The latter parameter is usually set to the VoIP codec frame inter-arrival time, e.g., every fourth frame.

There is also a small recovery mechanism that handles lost UL-MAP messages that prevents the SS from transmitting in the uplink. If the uplink request size R_{size} starts grow, the scheduler analyze the uplink request value and temporarily doubles or even triples the size of the UL grant.

More details on the implemented scheduler and support for other scheduling classes are given in [10].

2.2 Sub-MAP Messages

In order to reduce the data bandwidth overhead in sending DL-MAP or UL-MAP, the allocation of data bursts can be achieved by utilizing different types of messages that can be transmitted using more efficient modulation coding schemes (MCSs). The 802.16 standard introduces sub-MAPs that allow for splitting a MAP message into a number of independent messages, each of which is encoded with a more efficient MCS. Sub-MAPs follow the compressed MAP as shown in Fig. 1. The sub-MAP format is quite similar to the compressed MAP with a few additional enhancements, such as the CRC-16 field instead of CRC-32.

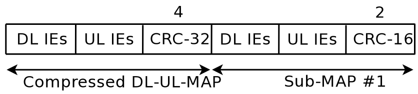


Fig. 1. Compressed MAP with sub-MAP

By using sub-MAPs, data burst allocation information can be transmitted more efficiently. The current version of the 802.16 standard supports up to three sub-MAP messages. To optimize burst allocation efficiency, different algorithms for dividing SSs and choosing an appropriate sub-MAP for data burst allocation can be used. For instance, SSs can be divided into different groups each assigned to a different sub-MAP based on their SINR (Signal to Interference-plus-Noise Ratio). As a result, signalling overhead for sending DL-MAP and UL-MAP is significantly reduced. The only drawback of the sub-MAP message is that if it is dropped, then both the DL-MAP and UL-MAP entries are lost. However, similarly to a compressed MAP message, the sub-MAP should be transmitted with quite a robust MCS to ensure that all the stations receive it correctly. On the other hand, sub-MAPs with a more robust MCS will send less data when compared to other sub-MAPs with a more efficient MCS. Thus, choosing an optimal MCS is a tradeoff between robustness and efficiency.

The 802.16 specification just defines the maximum number of sub-MAPs that can appear in a frame. The exact number, configuration, and MCS to encode a particular sub-MAP are left undefined. We have implemented a simple yet efficient algorithm to build sub-MAPs based on the number of bursts the BS scheduler allocates in a frame. In a few words, it starts a sub-MAP on a particular MCS if that MCS has the biggest number of Information Elements (IEs). Thus, each MCS can be assigned a weight that determines a preference for a sub-MAP message:

$$w_{\mathrm{MCS}} = \frac{\sum S_i^{\mathrm{IE}}}{S^{\mathrm{slot}}}, \tag{2}$$

where S_i^{IE} is the IE size measured in bytes and S^{slot} is the slot size that corresponds to the MCS, for which a weight is calculated. Since the weight determines only a preference for a particular MCS where we might build a sub-MAP, we need an iterative algorithm, a simplified form of which is shown in Fig. 2. The algorithm comprises the following stages:

1. Calculate a weight value for each MCS. Based on the number of uplink and downlink IEs, i.e., the number of data bursts that are encoded with the given MCS, each MCS is assigned a weight value that determines its preference for a sub-MAP candidate. To simplify further manipulations, the list is sorted based on the weight values in the descending order.
2. Consider an MCS as a candidate for a sub-MAP. The algorithm takes the (next) MCS with the highest weight value, adds a sub-MAP, and estimates the total number of slots that a new sub-MAP configuration needs. If it reduces the total signalling overhead, then the MCS is saved into a permanent sub-MAP configuration. Otherwise, the MCS is just skipped. To determine

whether the (next) MCS with the highest weight value reduces the total signalling overhead, the algorithm compares the total number of slots needed to encode IEs with the previous sub-MAP configuration. If the overhead is smaller in terms of slots, then update the permanent sub-MAP configuration.
3. The algorithm works until the maximum number of sub-MAPs is constructed or until all the MCSs are processed.

More details on the sub-MAPs as well as a full description of the implemented algorithm are given in [13].

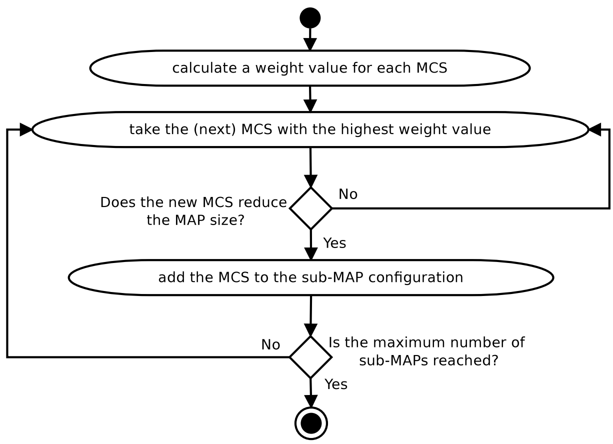


Fig. 2. Sub-MAP construction algorithm

3 Simulations

This section presents the dynamic simulation results of sub-MAPs with the VoIP application in the 802.16 network. To run simulations, we have implemented the 802.16 MAC and PHY levels in the NS-2 simulator [11]. The MAC implementation contains all the main features of the IEEE 802.16 standard, such as downlink and uplink transmission, connections, MAC PDUs, packing and fragmentation, the contention and ranging periods, the MAC level management messages, and the ARQ mechanism. The HARQ implementation supports Type I, i.e., chase combining (CC) [2].

To speed up simulations, we do not model all the PHY details but rather rely upon the effective SINR trace files taken from the system level simulator, where we modeled 19 cells with 3 sectors per each cell. This provides realistic wireless channel variations, which cause packet errors and drops. The relevant parameters are given in Table 1. All the other PHY mechanisms, such as channel measurements, channel reporting, link adaptation, and scheduling are present and explicitly modeled in the simulation environment.

Fig. 3 shows the network structure we used in the simulations. We concentrate on a single 802.16 sector with the BS and a number of SSs. The 802.16 network

Table 1. System level parameters

Parameter	Value
Reuse factor	1/3
Inter-site distance	1.5 km
Path loss model	UMTS 30.30
Slow fading std.	8 dB
Fast fading	Ped B (60%), Veh A (40%)
Antenna technique	SISO (1x1)
Antenna pattern BS/MS	3GPP / Omnidirectional
Antenna height BS/MS	32 / 1.5 m
Tx power BS/MS	20 / 0.2 W

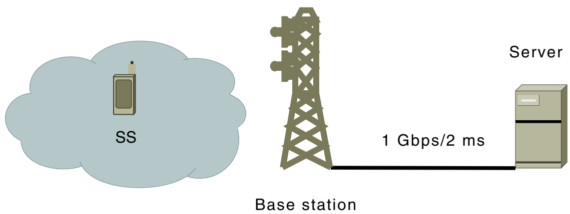


Fig. 3. Network structure

parameters are presented in Table 2[1]. Even though we study a symmetrical VoIP service, we choose the TDD DL/UL ratio in such a way that more resources are allocated to the DL sub-frame where signalling messages reside. While modeling the DL signaling messages, such as MAPs and sub-MAPs, we account for the fact that they can be dropped and apply the same error model as we do for normal data PDUs. As a result, if an SS looses a MAP or a sub-MAP message, it cannot receive or transmit data. It allows us to study how sub-MAPs impact performance of the VoIP service.

The BS runs the throughput-fair scheduling algorithm, general description of which is presented in [10]. In section 2.1, we gave a more detailed description of the VoIP scheduler. The link adaptation mechanism ensures the target FEC BLER of 10^{-1} for the HARQ-enabled connections.

Each SS establishes a basic management connection to exchange management messages with the BS. In addition, each SS has a bi-directional VoIP transmission with the server carried over two ertPS connections. The VoIP application parameters given in Table 3 are taken from the IEEE 802.16m Evaluation Methodology Document [3]. However, to stress the system, we model the worst case scenario, where each VoIP SS has the constant active bi-directional VoIP transmission during each simulation run. In addition, we limit the VoIP connection queue length to 3 SDUs to meet the air interface transmission delay of less than 100 ms. The ertPS connections use HARQ as a retransmission mechanism because ARQ cannot always ensure required delay limits. Furthermore, we turn the HARQ PDU

[1] These parameters conform to the WiMAX Forum mobile system profile [4].

Table 2. 802.16 network parameters

Parameter	**Value**
PHY	OFDMa
Bandwidth	10 MHz
Duplexing mode	TDD
Frames per second	200
Cyclic prefix length	1/8
TTG+RTG	296+168 PS
OFDM symbols	48
DL/UL symbols	28/19
DL/UL subcarrier alloc.	DL PUSC/UL PUSC
DL/UL slots	420/210
Channel report type / interval	CQICH / 20ms
Channel measurements DL/UL	preamble / data burst
Channel measurements filter	EWMA , $\alpha = 0.25$
Compressed MAPs	ON
sub-MAPs / max. number	ON / 3
Ranging transm. opport.	2
Ranging backoff start/end	1/15
Request transm. opport.	4
Request backoff start/end	2/15
CDMA codes	256
ranging+periodic ranging	64
bandwidth request	192
handover	–
HARQ	Type I (CC)
HARQ channels	16
HARQ buffer size	2048 B (per channel)
HARQ shared buffer	ON
HARQ max. retransmissions	4
HARQ ACK delay	1 frame
HARQ PDU SN	OFF
Fragmentation/packing	OFF

sequence numbering off to cater for smaller delays[2]. In addition, we disable both packing and fragmentation for VoIP, which in conjunction with a proper VoIP scheduler ensures absence of unnecessary MAC level overhead. Uplink ertPS connections use the CDMA contention as the resumption mechanism to inform the BS that there is an uplink data to be transmitted. As simulated in [9], this is a more efficient way when compared to polling.

Extensive simulations were conducted to study VoIP capacity of the whole system. For these purposes, we varied the number of VoIP connections in the system. A different approach is to inject a large number of VoIP connections and then let the admission control module to drop connections if the network capacity is exceeded [8]. Since we do not aim at studying the admission control module, we just disabled it.

For each simulation case, we made 32 simulation runs to obtain proper confidence intervals. Each simulation run lasted for 13 seconds, where the actual

[2] The HARQ PDU ordering is usually turned on for the TCP connections where the TCP receiver may react unpredictably for packets arriving in the wrong order. In case of VoIP, a receiver usually implements a so-called de-jitter buffer, depth of which varies from 20 ms to 60 ms. An alternative VoIP configuration is to enable PDU SN and tune the HARQ ordering buffer timeout based on the maximum number of HARQ retransmissions.

Table 3. VoIP parameters

Parameter	Value	
Codec	G.729	
Active state	100%	
Aggregation interval	20 ms	
Voice Payload	20 bytes	
802.16 Generic MAC header HARQ CRC	6 bytes 2 bytes	
IP/UDP/RTP headers	40 bytes (no PHS)	2 bytes (PHS)
Total VoIP packet size	68 Bytes	30 Bytes

data transmission starts at the 3rd second because the first SSs have to enter the cell and register at the BS.

3.1 Performance Criteria

In this sub-section we describe two main VoIP performance criteria that we used: delays and packet loss ratio.

Delays. The VoIP SDU delay is defined as

$$Delay = T^{dep} - T^{arr}, \tag{3}$$

assuming that the packet destined for SS arrives at the BS(SS) at time T^{arr} and is delivered to the SS(BS) at time T^{dep}.

Following [3], we check that 98% of connections deliver 98% of VoIP packets within the delay bound of 100 ms. For these purposes, first for each connection we determine percentage of packets that meet the delay bound of 100 ms. Then, these values are used to build another CDF (cumulative distribution function) that shows how many connections meet the target requirements.

Packet Loss Ratio. According to [3], the packet loss ratio per connection is defined with the following criterion:

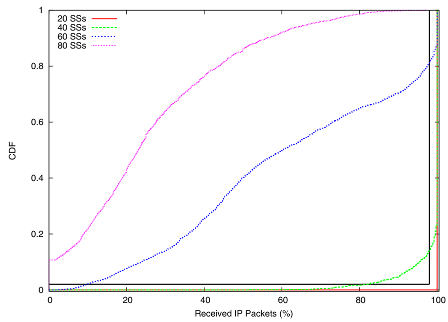
$$R^{loss} = 1 - \frac{P^{delivered}}{P^{total}}, \tag{4}$$

where $P^{delivered}$ includes the total number of successfully delivered packets and P^{total} includes packets that were transmitted over the air interface and packets that were dropped prior to transmission. If more than 2% of the VoIP packets are dropped, erased or not delivered successfully to the SS within the delay bound of 100 ms, then SS is not satisfied.

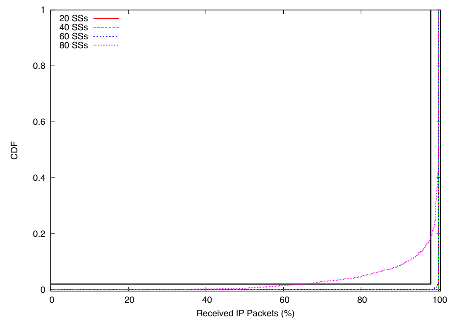
In modeling traffic from delay sensitive applications, packets may be dropped if packet transmissions are not completed within a specified delay bound. The impact of such dropped packets can be captured in the packet loss rate.

3.2 Delays

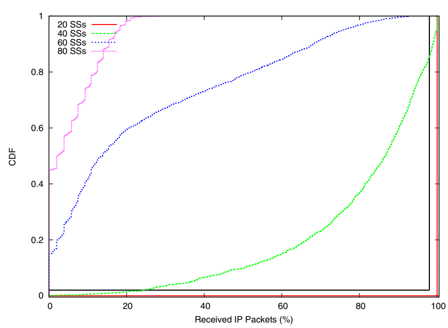
In this sub-section we analyze how injecting a different number of VoIP connections and enabling sub-MAPs impact the VoIP capacity and associated transmission delays. To see the cases when SSs do not meet their delay requirements, we introduce boundary lines for X and Y - axis that show whether 98% of packets are delivered successfully to the 98% SSs within the delay bound of 100 ms. We consider separately a case with disabled sub-MAPs and enabled sub-MAPs.



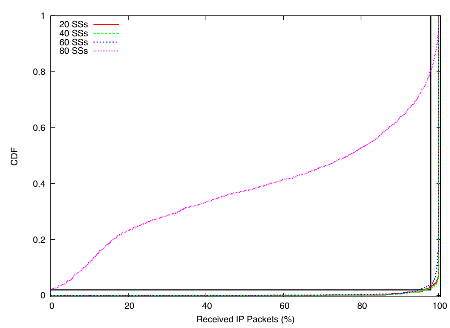
(a) disabled sub-MAPs, DL



(b) enabled sub-MAPs, DL



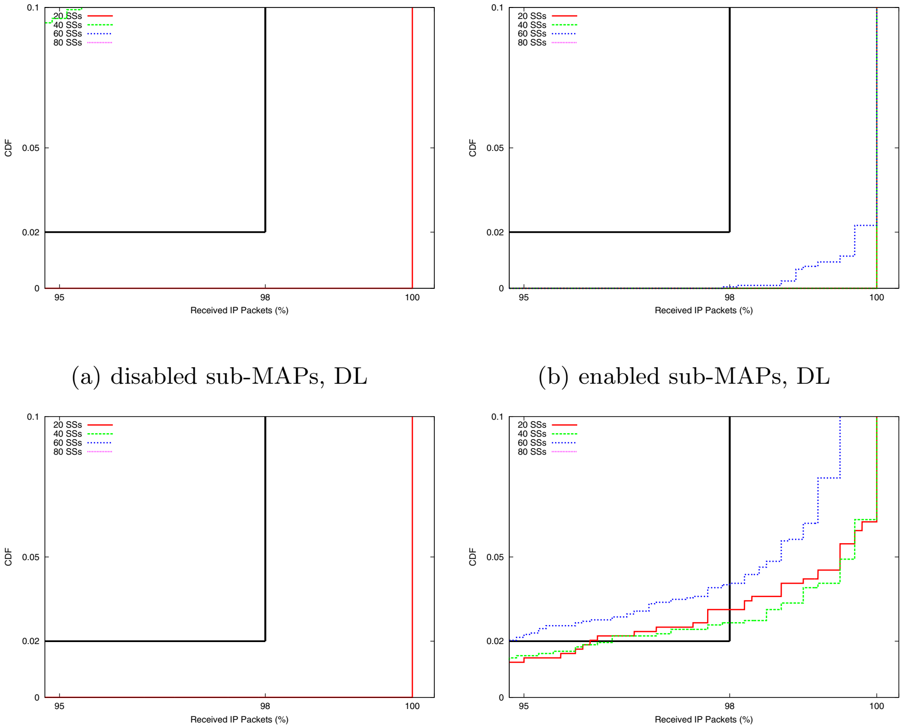
(c) disabled sub-MAPs, UL



(d) enabled sub-MAPs, UL

Fig. 4. VoIP SDU E2E delay CDF

Results with end-to-end delay CDFs are presented in Fig. 4(a) and Fig. 4(c) for the downlink and uplink directions respectively. Fig. 5(a) and Fig. 5(c) demonstrate the same case with more detailed elaboration for the downlink and uplink parts respectively. Fig. 4(a) shows that if sub-MAPs are disabled then the downlink VoIP capacity is only 20 SSs, whereas a bigger number of SSs start to experience downlink delays larger than 100 ms. As can be seen from Fig. 4(c), the uplink capacity is also limited by 20 SSs.



(a) disabled sub-MAPs, DL

(b) enabled sub-MAPs, DL

(c) disabled sub-MAPs, UL

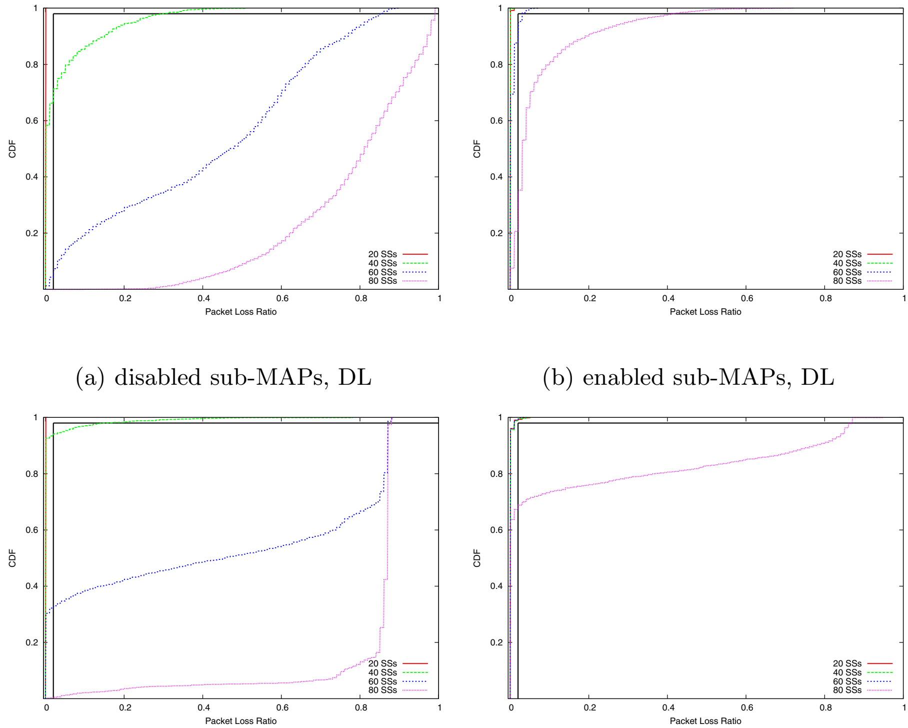
(d) enabled sub-MAPs, UL

Fig. 5. VoIP SDU E2E delay CDF

Next, we enable sub-MAPs and analyze the upper limits for the VoIP capacity. The results are shown in Fig. 4(b) and Fig. 4(d) for the downlink and uplink directions. When compared to Fig. 4(a), where the VoIP capacity is limited by 20 SSs in the downlink direction, Fig. 4(b) illustrates clearly that enabled sub-MAPs improve overall VoIP capacity up to 60 VoIP connections. In the uplink, as can be seen in Fig. 4(d), the VoIP capacity is around 60 SSs.

Fig. 5(b) and Fig. 5(d) show more detailed elaboration for the downlink and uplink parts respectively. Having compared Fig. 5(a) and Fig. 5(b), it is noticeable that enabled sub-MAPs start to bring more gain.

Fig. 5(d) also presents eloquently how enabled sub-MAPs tend to increase transmissions delays for a small number VoIP connections. When compared to Fig. 5(c), even 20 VoIP connections have slightly larger delays. The reason for this phenomena is dropped sub-MAP messages. If it is dropped, then both the DL-MAP and UL-MAP entries are lost. If an SS fails to receive the sub-MAP message, it cannot transmit anything in the UL direction thus causing queue growths. It takes some time for the BS scheduler to detect this situation and allocate a larger UL grant.



(a) disabled sub-MAPs, DL

(b) enabled sub-MAPs, DL

(c) disabled sub-MAPs, UL

(d) enabled sub-MAPs, UL

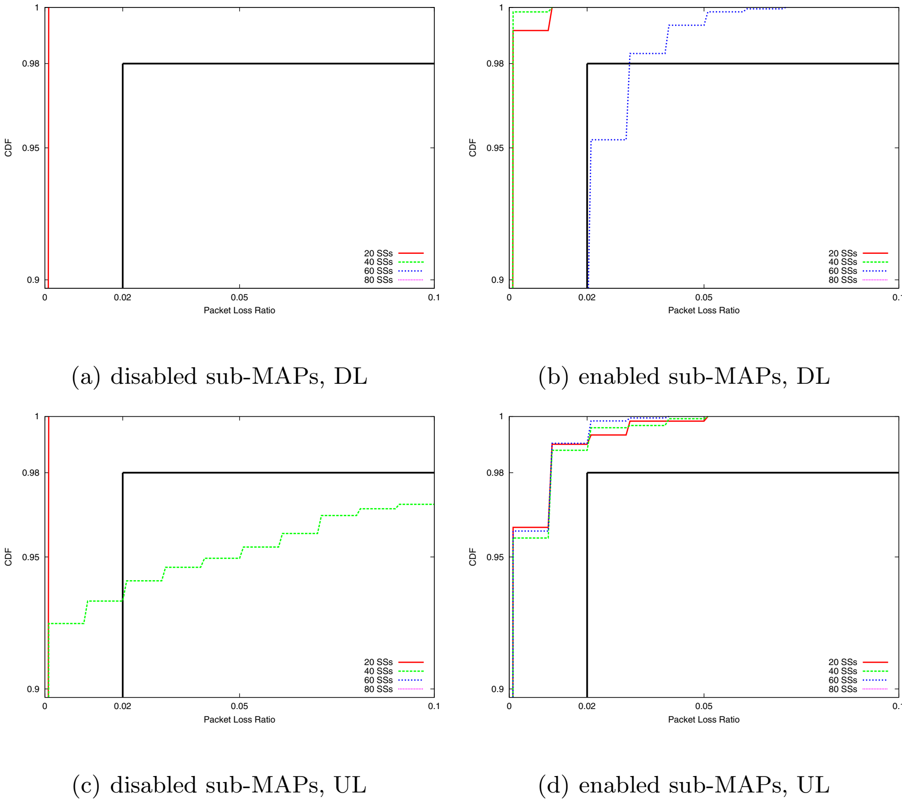
Fig. 6. VoIP Packet Loss Ratio

3.3 Packet Loss Ratio

In this sub-section we analyze how different numbers of VoIP connections and sub-MAPs affect the VoIP capacity and the packet loss ratio. As in the case of delay analysis, we also apply boundary lines for X and Y - axis to ensure that 98% of SSs experience packet drops of less than 2%.

The results for packet loss ratio CDFs with disabled sub-MAPs are presented in Fig. 6(a) and Fig. 6(c) for the downlink and uplink parts, while Fig. 7(a) and Fig. 7(c) demonstrate more detailed elaboration for the downlink and uplink parts respectively. As seen from Fig. 6(a) and Fig. 6(c), the only case with 20 SSs shows the packet loss ratio of less than 2% of both the downlink and uplink directions, while all the other cases exceed the target packet loss ratio due to frequent packet drops caused by queue overflows.

As we enable sub-MAPs, we can decrease drastically VoIP packet drops because now the system has more resources to serve VoIP connections. As can be seen from Fig. 6(b) and Fig. 7(b), almost 60 DL VoIP connections can be served. As for the uplink, as can be seen from Fig. 6(d) and Fig. 7(d), enabled sub-MAPs



(a) disabled sub-MAPs, DL

(b) enabled sub-MAPs, DL

(c) disabled sub-MAPs, UL

(d) enabled sub-MAPs, UL

Fig. 7. VoIP Packet Loss Ratio

decrease the packet loss ratio up to 60 SSs, but cannot meet the defined 2% with 80 SSs.

4 Conclusions

This paper contributes to evaluating the VoIP capacity with enabled sub-MAPs in the IEEE 802.16 OFDMa system. We ran complex dynamic simulations with all the major radio resource management algorithms of the IEEE 802.16 system and the VoIP service. The results demonstrate that it is reasonable to enable sub-MAPs to support more VoIP connections. Otherwise, the signalling overhead, as a part of the 802.16 air interface, creates a bottleneck for the 802.16 OFDMa system. In particular, our simulation results show the significant VoIP capacity improvement. However, the obtained performance with enabled sub-MAPs may vary depending on the network parameters and traffic load.

It is worth mentioning that sub-MAPs must be transmitted with quite a robust MCS to ensure that all the SSs receive them correctly. Otherwise, a loss of the downlink sub-MAP results in a situation when the BS reschedules the same data because an SS does not send a positive HARQ ACK message. In the uplink direction, if an SS cannot decode a sub-MAP then it fails to transmit any data. As a result, uplink queues start to grow thus increasing the transmission delays. As a possible solution, uplink sub-MAP messages may be encoded with a more robust MCS when compared to the downlink direction.

An interesting outcome of these dynamic simulations is that sub-MAPs have a negative impact on the UL channel estimation. If the sub-MAP encoding an UL allocation is lost, then an SS does not transmit thus preventing the BS from estimating the UL channel state that varies due to the fast fading and SS movements. A possible solution is a so-called UL sounding defined in the 802.16 specification.

Our further research will aim at studying how sub-MAP messages can be used in conjunction with persistent allocations. It is anticipated that even more VoIP connections can be admitted. On the other hand, a static nature of persistent allocations and a higher error rate of sub-MAPs may cause performance degradation of the VoIP service.

Acknowledgments

We express our acknowledgments to Alexander Sayenko (Nokia Siemens Networks) and Olli Alanen (Magister Solutions) who provided valuable comments on the simulation setup and results. We also would like to thank Henrik Martikainen, Oleksandr Puchko and Vesa Hytönen (University of Jyväskylä / Telecommunication laboratory) for their contribution to the implementation of the 802.16 MAC and PHY levels in the NS-2 simulator. Special thanks to Timo Hämäläinen (University of Jyväskylä / Telecommunication laboratory) for supervising this work.

References

1. Air interface for fixed broadband wireless access systems. IEEE Standard 802.16 (June 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 (December 2005)
3. IEEE 802.16m evaluation methodology document (EMD). IEEE 802.16 Broadband Wireless Access Group. (March 2008)
4. WiMAX Forum Mobile System Profile, Release 1.0 Approved Specification, Revision 1.6.1 (April 2008)
5. Air interface for broadband wireless access systems. IEEE Standard 802.16 (Rev. 2) (May 2009)
6. Fong, M.-h., Novak, R., Mcbeath, S., Srinivasan, R.: Improved VoIP capacity in mobile WiMAX systems using persistent resource allocation. IEEE Communications Magazine 46(10), 50–57 (2008)
7. Gross, J., Geerdes, H.-F., Karl, H., Wolisz, A.: Performance analysis of dynamic OFDMA systems with inband signaling. IEEE Journal on Selected Areas in Communications 24(3), 427–436 (2006)

8. Lakkakorpi, J., Sayenko, A.: Measurement-based connection admission control methods for real-time services in IEEE 802.16e. In: Second International Conference on Communication Theory, Reliability, and Quality of Service, CTRQ 2009, pp. 37–41 (July 2009)
9. Lakkakorpi, J., Sayenko, A.: Uplink VoIP delays in IEEE 802.16e using different ertPS resumption mechanisms. In: Third International Conference on Mobile Ubiquitous Computing, Systems, Services and Technologies, pp. 157–162 (October 2009)
10. Sayenko, A., Alanen, O., Hämäläinen, T.: Scheduling solution for the IEEE 802.16 base station. Computer Networks 52, 96–115 (2008)
11. Sayenko, A., Alanen, O., Martikainen, H., Tykhomyrov, V., Puchko, O., Hämäläinen, T.: WINSE: WiMAX NS-2 Extension. In: 2nd International Conference on Simulations Tools and Techniques (March 2009)
12. So, J.-W., Maetan-dong, Yeongtong-gu, Suwon-si, Gyeonggi-do: An efficient uplink mapping scheme for VoIP services in the IEEE 802.16e OFDMA system. AEU - International Journal of Electronics and Communications 62, 768–776 (2008)
13. Tykhomyrov, V., Sayenko, A., Puchko, O., Hämäläinen, T.: Decreasing the MAP overhead in the IEEE 802.16 OFDMA system. In: European Wireless Conference (2010) (accepted for publication)
14. Yeom, J.-H., Lee, Y.-H.: Efficient transmission of multicast maps in IEEE 802.16e. IEICE Transactions 91-B(10), 3157–3161 (2008)