On the Performance Gains of VoIP Aggregation and ROHC over a WirelessMAN-OFDMA Air Interface

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***Abstract*—A growing number of mobile WiMAX deployments are in progress world-wide and the technology is anticipated to play a key role in next generation mobile broadband wireless networks. However, although the theoretical potential of WiMAX technologies is already well established, independent, publicly released, thorough evaluations of WiMAX network performance in the real world are yet to become available. In this study, we consider synthetic Voice over IP (VoIP) performance over the WirelessMAN-OFDMA air interface of a state of the art mobile WiMAX testbed operating at the 3.5 GHz frequency band and quantify the benefits of employing VoIP aggregation and Robust Header Compression (ROHC). Although VoIP aggregation and ROHC have been proposed and thoroughly evaluated through simulation and modeling, this is the first study to present empirical results from employing them over the WirelessMAN-OFDMA air interface of a real-world WiMAX system. Our results indicate that the combined use of VoIP aggregation and ROHC can increase the number of flows sustained without loss by approximately a factor of three.**

I. Introduction

Worldwide Interoperability for Microwave Access (WiMAX) is currently one of the leading contestants in the heating beyond 3G technology race. In principle, WiMAX can meet the strict requirements posed by 4G, delivering high data rates, full mobility support, and adequately low link latencies. WiMAX proponents argue that deployments will clear the hype that has surrounded WiMAX for several years.

WiMAX technology is based on the family of standards put together by the IEEE 802.16 Working Group (WG). More specifically, WiMAX uses PHY and MAC layer configurations which are subsets of the specifications presented in [1]. These subset configurations are specified by the WiMAX Forum in the WiMAX system profiles. All WiMAX Forum certified equipment is tested against these specifications with certification profiles which are working configurations drawn from the actual system profiles. Currently, manufacturers can test their WiMAX equipment either against a fixed or a mobile WiMAX system profile. In the fixed WiMAX system profile, the WirelessMAN-OFDM air interface can be used either in Time Division Duplex (TDD) or Frequency Division Duplex (FDD) mode. In the mobile WiMAX system profile the WirelessMAN-OFDMA air interface is employed by using TDD. WiMAX equipment certified according to the mobile system profile must support both fixed and mobile subscribers.

Voice transmission over packet switched networks has been a hot topic in the telecommunication and networking research already for several years, as Voice over IP (VoIP) is commonly foreseen as one of the key applications for wireless networks. From the network operator's point of view this means that 4G technologies should be able to provide high capacity for VoIP services alongside other data packet applications. As VoIP applications usually exchange large quantities of small data packets over the network, as well as make high and strict demands on one-way end-to-end delays and jitter in order to deliver good conversation quality, the network technologies in use should be able to handle such traffic efficiently in both fixed and mobile environments.

The special requirements of VoIP have also been taken into consideration in the IEEE 802.16 WG, as the ongoing development of the standards includes VoIP delivery issues at least in two separate task groups. Focusing on the near future, the Maintenance Task Group published the standard IEEE 802.16-2009 [1] in May 2009. This new revision consolidates the previous IEEE 802.16-2004 standard with its amendments and corrigendums into the same document. The new standard also enables VoIP capacity improvements by introducing persistent air interface resource assignments and, thus, reduces the control signaling overhead caused by dynamic scheduling activities. In addition, Task Group m is working on amendment IEEE 802.16m [2], which will introduce an improved air interface capable of supporting higher data rates and lower link latencies than the one currently specified in the standards, aiming to fulfil the requirements of the International Mobile Telecommunications (IMT) -Advanced systems [3], specified by the International Telecommunication Union's Radiocommunication Sector (ITU-R). The IEEE 802.16m amendment is still in the pre-draft stage and is currently expected to be finalized in late 2009 - early 2010 [4].

In this paper both state of the art IEEE 802.16 compliant technology and VoIP are evaluated, as mobile WiMAX equipment based on [5] and [6] is put into the test and its performance under VoIP traffic is measured and analyzed. Hence, the main contribution of the paper is one of the first publicly available empirical multimedia streaming performance evaluations of a WiMAX testbed using the WirelessMAN-OFDMA air interface at the 3.5 GHz frequency band, as voice sample aggregation and Robust Header Compression (ROHC) are tested separately and in unison as performance enhancement techniques over the testbed. As an additional contribution, a performance comparison between typical real-world Orthogonal Frequency Division Multiplexing (OFDM) and Orthogonal Frequency Division Multiple Access (OFDMA) WirelessMAN air interface deployments is presented.

The rest of this paper is organized as follows. Section II briefly overviews related work. Section III introduces the testbed, methods, and tools used in the measurements and Section IV presents our measurement results. Section V discusses further the reported results and, finally, Section VI concludes this paper and outlines future work items.

II. Related Work

To the best of our knowledge, the only publicly available mobile WiMAX measurement study is [7], in which Kim et al. inject User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) flows and quantify link capacity, network throughput and round trip times (RTT). The study is performed over a public / commercial WiBro network in Seoul, South Korea, operating at the 2.3 GHz frequency band and using 10 MHz bandwidth. Kim et al. find that the average RTT ranges 134-176 ms, depending on the location of the WiBro Mobile Station (MS). Maximum UDP goodput for a stationary MS is 10.6 Mb/s for the Downlink (DL) and 3.1 Mb/s for the Uplink (UL). For a non-stationary MS, UDP goodput drops to 6.5 Mb/s and 1.2 Mb/s in DL and UL, respectively. Although these measurements are done in a different frequency band and with a different bandwidth allocation, the results in [7] provide valuable comparison material for our own measurements.

So [8] studies VoIP performance in an OFDMA DL based on [6]. Using modeling, and taking into consideration the MAP message overhead in the DL frames, So determines DL capacity for single and multiple VoIP clients. Theoretical performance curves are presented for mean throughput and packet dropping probability for a number of VoIP users with different modulation and coding schemes. Neither [7] nor [8] consider the effect of employing VoIP aggregation or ROHC.

This paper also adds to our own WiMAX measurement studies performed at the VTT Converging Networks Laboratory. For example, in previous work we quantified the gains from VoIP packet aggregation in a WirelessMAN-OFDM air interface using a fixed WiMAX link compliant with the IEEE 802.16-2004 standard [9]. The measurements pointed to substantial gains in the number of simultaneous synthetically-generated G.723.1-encoded VoIP flows sustained by the wireless link. By aggregating two voice samples into a single VoIP packet at the application layer, the fixed WiMAX testbed could sustain roughly twice the amount of simultaneous one-way VoIP flows. Capacity gains are even larger when three voice samples are aggregated into a single VoIP packet.

More recently, we measured the performance of G.729.1-encoded VoIP traffic over the same fixed WiMAX testbed and explored the potential benefits of ROHC [10]. Contrary to our expectations we found that ROHC alone does not lead to significant gains in capacity. However, when two voice samples are aggregated into a single VoIP packet at the application level and ROHC is employed at the lower layers, the capacity gains can exceed 80%. In this paper, we follow the methodology introduced in [10] and measure the gains from VoIP aggregation and ROHC in our mobile WiMAX testbed. The aim of this paper is to add to our understanding of the real-world potential of these technologies and provide further measurement data about mobile WiMAX systems employing WirelessMAN-OFDMA. To this extent, [7] and [10] provide good comparison points as we investigate the effectiveness of these techniques with a different WiMAX PHY configuration.

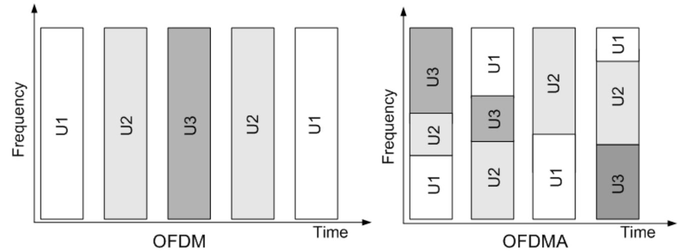

Fig. 1. OFDM and OFDMA transmission schemes

III. Methodology

Fig. 1 is a simplified illustration of the main difference between the OFDM, typically used in current fixed WiMAX implementations, and OFDMA which is used in mobile WiMAX. OFDMA basically implements multiple access capabilities on top of OFDM and allows different users (U1, U2, and U3 in Fig. 1) to access the channel simultaneously by dividing the available bandwidth into sub-carriers, as in OFDM, and grouping adjacent sub-carriers into blocks called sub-channels. By allocating resources to users on sub-channel by sub-channel basis, the transmission power, modulation scheme, and error correction coding can be adapted to prevailing channel conditions in each transmission time slot. During the measurements presented in this paper, the mean Received Signal Strength Indication (RSSI) and the mean Carrier to Interference and Noise Ratio (CINR) values were -41 dBm and 30 dB, respectively.

A. Testbed Description

Fig. 2 illustrates the mobile WiMAX testbed used in this measurement study. It comprises an Alcatel-Lucent 9116 BS, a ZyXEL MAX-210M1 Customer Premises Equipment (CPE) and two GNU/Linux PCs, connected to the testbed on the Radio Access Network (RAN) and Core Network (CN) side with Gigabit Ethernet. RAN uses Alcatel-Lucent's software release W3.0. Synchronization of the end-hosts is carried out with a server running PTPd [11], which is an open source implementation of the Precision Time Protocol (PTP) [12]. PTPd allows synchronization of Commercial of the Shelf (COTS) PC clocks at the end hosts with an accuracy of tens of microseconds, as detailed in [13]. The PTP server connects to the hosts via separate Ethernet interfaces; the synchronization traffic does not interfere with the measurement traffic.

The key parameters of the air interface are also shown in Fig. 2. UL modulation is restricted to 16 QAM by the CPE, which limits the maximum achievable UL throughput in our testbed. In addition, the CPE only supports 1/2 Forward Error Correction (FEC) coding rate. The BS can support 64 QAM (FEC: 3/4) for both DL and UL. MAC scheduling is best effort. In addition, the equipment implements a TDD duplexing method on its OFDMA PHY layer. We fix the DL/UL ratio to 2:1 for all measurements presented herein as it is the default configuration for the used equipment.

B. Measurement Scenarios

In application layer aggregation, multiple voice samples are placed in a single VoIP packet before it is RTP-encapsulated [14]. Application layer aggregation is allowed by the RTP specification and it is currently supported by many VoIP applications. In this study, voice sample aggregation is emulated at the application layer.

ROHC [15] compresses, among others, the RTP, UDP, and IP headers capitalizing on their redundancy as the headers contain fields that change rarely or never during the lifetime of a traffic flow. These fields include source and destination addresses in the IP header and port information in the UDP header. By utilizing the predictability and dependencies of other fields, these can be removed from most of the packets sent over a point-to-point link.

The ROHC RTP profile is emulated by compressing RTP/UDP/IP headers (12/8/20 bytes, respectively) to 4 bytes only. We emulate the unidirectional mode (U mode) where no feedback channel between a sender and a receiver exists. In the U mode, the decompression side must individually maintain the context of each sender side. Context is initialized in the beginning of a session and is periodically refreshed by the sender. In this study, however, the compression rate is fixed and context initialization and refreshment is not emulated.

ITU-T G.729.1 [16] is a scalable wideband codec widely supported by modern VoIP applications, which features a layered structure and support for different audio quality levels. The core layer, corresponding to 20 ms of voice, is based on the ITU-T G.729 [17] narrowband codec; the G.729.1 extension layers provide for scalable wideband speech and audio compression capabilities.

The layered structure of G.729.1 allows for 12 different operational bitrates ranging 8-32 kb/s. We opted to use a four-layer coding, which produces a voice payload of 40 bytes and an operational bitrate of 16 kb/s [18]. Fig. 3 summarizes the different packet sizes used in the study for plain and aggregated VoIP, with and without ROHC Compressed Headers (CH). Note that the adaptive features of G.729.1 are not covered in this study but are part of our research agenda.

All VoIP traffic is synthetically generated using a traffic generator implemented in Perl, as the available traffic generators tested before the study did not fulfill all the accuracy requirements for these measurements. Because the inaccuracy is caused by the operating system process scheduling, not the traffic generators themselves, we implemented our own traffic generator using adaptive sleep times between injected packets. With this approach, the maximum inaccuracy remains constantly under 0.4%. Each measurement run is configured

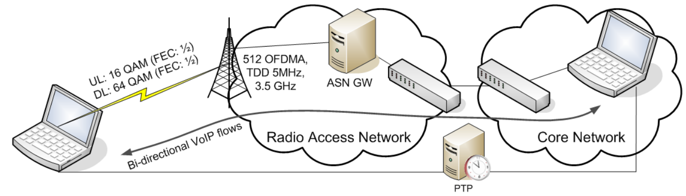

Fig. 2. Schematic of the mobile WiMAX testbed

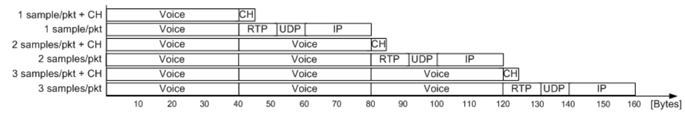

Fig. 3. Packet sizes used in different measurement cases

to run for 100 s and with the custom-made traffic generator used in this study, none of the runs lasted more than 100.4 s.

IV. Results

In this section we report on the link capacity of the testbed and evaluate VoIP over the WirelessMAN-OFDMA air interface of a state of the art mobile WiMAX testbed, measuring packet loss, goodput and one-way delay.

A. Effective Link Capacity

Before proceeding with our synthetic VoIP evaluation, we performed several application-layer measurements in order to determine the maximum capacity of the WiMAX link, in practice. First, we used *pchar* [19], an open source bandwidth estimation tool over the empty WiMAX link. *pchar* measures the characteristics of a network path using UDP probe packets. Routers along the network path respond to the probe packets using Internet Control Message Protocol (ICMP), and based on these responses, *pchar* estimates the RTT, available bandwidth, and average queuing delay for every hop in the network path.

However, *pchar* which was originally designed for wired networks, fails to provide accurate estimates for WiMAX link capacities. Certain WiMAX MAC and PHY layer features as well as the link latencies and jitter, which are several orders of magnitude larger than in Ethernet, lead *pchar* to report available bandwidth figures which do not correspond to reality. The asymmetric nature of the WiMAX link plays a crucial role here. Asymmetry between DL and UL directions distorts estimations based on ICMP responses as latency and jitter are different depending on the direction that the packet travels in the WiMAX link. As a result, *pchar* misestimates both DL and UL capacity. Table I summarizes the range of the *pchar* measurement results.

Since we could not obtain reliable estimates using *pchar*, we measured the DL and UL capacity with Constant Bit Rate (CBR) traffic using different UDP packet sizes. The results from these baseline measurements are also given in Table I. The measurements clearly show that the WiMAX DL performs particularly badly when small packets are transmitted over the wireless link. This may suggest some buffer underprovisioning in the outbound interfaces of the BS, or that small packets

TABLE I
ESTIMATED AND MEASURED MOBILE WIMAX LINK CAPACITIES

	DL	UL
pchar estimate (Mb/s)	min: 5.88; max: 16.85	min: 0.36; max: 4.40
Packet size (bytes)	Mean Effective Capacity (Mb/s)	
128	1.72	1.46
500	4.66	1.49
1400	5.13	1.53

cannot be handled in an expedient manner. For example, if we use CBR traffic with 128-byte long packets, the effective capacity is only 1/3 of that measured with CBR traffic with 1400-byte long packets. The term “effective capacity” here refers to the amount of traffic we can inject in the WiMAX link while observing negligible packet loss. On the contrary, the UL effective capacity is marginally affected when we use different packet sizes, and is approximately 1.5 Mb/s.

B. VoIP Packet Loss and Goodput

We now proceed with the synthetic VoIP measurements. Fig. 4 illustrates the mean VoIP packet DL loss rates as a function of the number of simultaneous bidirectional VoIP flows injected in the mobile WiMAX testbed. When neither voice sample aggregation nor ROHC is used, the DL can sustain 18 simultaneous VoIP conversations before the packet loss rate exceeds the theoretical quality threshold of 5% specified in [20]. This translates into a mean throughput of just under 280 kb/s, as shown in Fig. 5, which is approximately only 5.5% of the maximum effective DL capacity as reported in Table I. We observe that ROHC alone does not improve performance significantly. Although ROHC can reduce dramatically the header size, it cannot alleviate the deficiency in handling large numbers of small packets from different sources.

When application-layer voice sample aggregation is employed, the performance of the WiMAX link improves substantially. By bundling two voice samples into a single VoIP packet, we find that the DL can sustain nearly twice the number of simultaneous VoIP conversations without quality impairments. Average packet loss rate remains under 5% with 34 simultaneous emulated VoIP conversations; this yields an average goodput of over 510 kb/s. Furthermore, because of the larger packet size, ROHC starts to register gains. With ROHC and VoIP aggregation, the WiMAX link can sustain 35 simultaneous VoIP conversations. This saturates the DL goodput to approximately 550 kb/s. If the number of the VoIP conversations is increased to 36, the average packet loss rate is just over 6%, which exceeds the theoretical quality threshold, but may still allow for decent-quality VoIP conversations.

By aggregating three voice samples into a single VoIP packet, the amount of sustained simultaneous VoIP conversations increases to 47 allowing for an average DL goodput of over 720 kb/s. With 48 simultaneous flows the average packet loss is approximately 6.2%, which exceeds the 5% threshold. If ROHC is employed, 48 simultaneous VoIP conversations can be sustained in the DL without quality degradation. If we inject one more bidirectional VoIP flow, the average packet

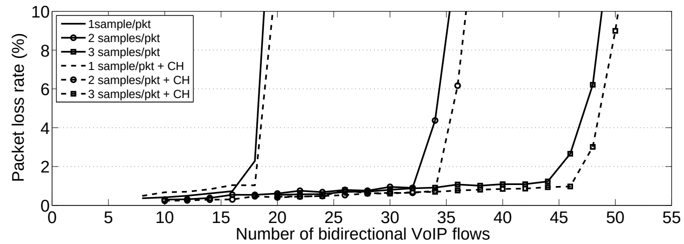

Fig. 4. Downlink packet loss rate

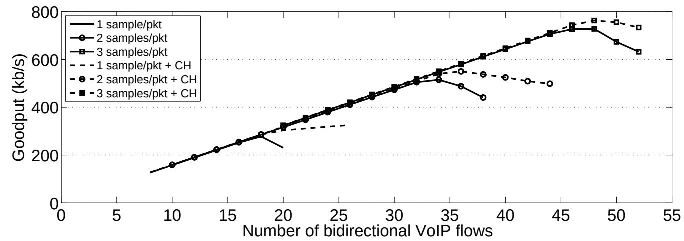

Fig. 5. Downlink goodput

loss nears 6%. The achieved average DL goodput with ROHC is just over 760 kb/s. This means that by using both voice sample aggregation and ROHC, the gains in VoIP call capacity are 89% and 161% for the aggregation of two and three voice samples, respectively, when compared to a non-aggregated case without ROHC. In short, the mobile WiMAX testbed can sustain twice or thrice as many bidirectional flows depending on the VoIP aggregation level and ROHC usage.

With respect to packet loss in UL, we note that it remains largely the same throughout the measurement campaign and does not exceed 1.5%. Unlike DL, the UL implementation seems to cope better with small packets. The minimum average packet loss rates were measured for VoIP packets carrying a single voice sample with ROHC headers (0.02%) and the maximum with VoIP packets carrying three voice samples without ROHC (1.4%). These results strongly indicate that the BS and CPE outbound interfaces are managed differently. Moreover, the UL was never saturated in any of these measurements, and we observe a linear increase in goodput for all test cases. In other words, although one would expect that in light of the capacity measurements reported above, the UL would be the bottleneck for bidirectional VoIP traffic, we see that in practice this is not the case. This very point emphasizes the need for thorough, third party empirical measurements of mobile WiMAX performance under various traffic mixes, which go beyond manufacturer and operator advertisements.

C. One-way Delay

From the VoIP user point of view, one-way delays exceeding 150 ms are expected to have a negative impact on the quality of the VoIP conversation as the human ear begins to notice the excess delay [20]. Clearly, voice sample aggregation at the application layer introduces buffering delays, which need to be carefully considered. For example, when aggregating two samples in a VoIP packet, every second sample is required

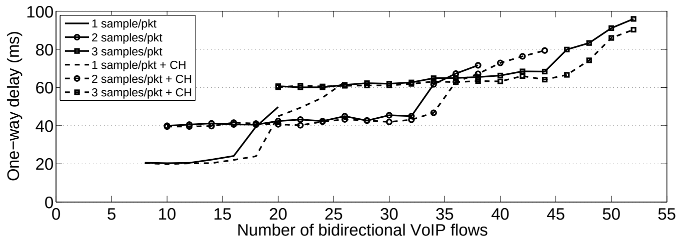

Fig. 6. Downlink one-way delay

TABLE II
PERFORMANCE COMPARISON BETWEEN OFDMA AND OFDM AS A NUMBER OF SIMULTANEOUS VOIP CONVERSATIONS

	1 sample/pkt + ROHC	2 samples/pkt + ROHC	Effective link capacity
OFDMA TDD, BW: 5 MHz	18	35	DL: 5.1 Mb/s UL: 1.5 Mb/s
OFDM FDD, BW: 3.5 MHz	69	121	DL: 9.4 Mb/s UL: 5.5 Mb/s
Difference	283%	246%	DL: 84.3% UL: 267%

to be buffered at least an additional 20 ms (the encoding delay of a G.729.1 codec). Hence, we decided that in practice aggregating four or more voice samples may lead to worse subjective performance from the end-user point of view. Aggregating three samples adds 40 ms of excess one-way delay to every third sample, as it is buffered waiting for the following two samples before it is transmitted. Despite the aggregation delays, Fig. 6 shows that the one-way link latencies in the DL do not exceed 100 ms under any of the tested configurations. Thus, in our mobile WiMAX testbed, packet loss, not one-way delay is the limiting factor for VoIP capacity.

Overall, one-way delays increase slightly as we inject more VoIP flows into the WiMAX link. One-way delays start to increase rapidly as we approach the DL saturation points. The saturation points are not the same for all configurations and are tightly correlated with the packet loss rates. Without ROHC, the saturation points correspond to 18, 34, and 47 simultaneous VoIP conversations when one, two, and three voice samples are aggregated into a single VoIP packet, respectively. With ROHC, the respective corresponding saturation points are 18, 35, and 48. When no aggregation is employed, one-way delay ranges between 20-30 ms. One-way delays were measured in the UL as well, and their range remained at 30-90 ms across all measurement configurations. In the UL, one-way delays range 30-35 ms when no aggregation is employed. As expected, the measured inter-packet one-way delays are higher when voice sample aggregation is employed and lower when no voice sample aggregation takes place.

D. OFDMA vs. OFDM

By using testbed deployments with default parameters for WirelessMAN links in WiMAX systems, a performance comparison between the OFDMA and OFDM air interfaces can be performed between the results presented in this paper and in [10]. Table II summarizes the VoIP performance and effective capacities of both configurations.

The fact that the modulation schemes and bandwidths available for DL and UL communication differ in the used configurations can also be seen from the measurement results. Even though the OFDMA TDD uses larger channel bandwidth, the simple OFDM FDD has a larger maximum throughput and it outperforms the more sophisiticated OFDMA TDD when delivering symmetric VoIP streams over the air interface. As the dynamic resource allocation in OFDMA increases the management overhead in the DL subframe channel descriptions and MAP messages even further when large quantities of small packets are traversing the link, the resulting difference in the performance is clearly visible in the measured capacity figures presented in Table II. Our results indicate that VoIP aggregation can be crucial in current deployments of both fixed and mobile WiMAX in order to significantly increase call capacity.

V. TOWARDS DYNAMIC VOIP AGGREGATION

As the measurement results with emulated traffic patterns in Section IV indicate, great gains can be achieved in the overall VoIP performance of systems based on the WirelessMAN-OFDMA air interface specified in [5] and [6], if the packet overhead is decreased with suitable techniques. On the other hand, as a basic feature of all wireless communication, the radio channels are bound to change over time and, thus, simply choosing one overhead mitigation method in the beginning of a VoIP session and keeping it fixed during a VoIP call may not be enough to guarantee optimized performance. The situation naturally becomes even more difficult when user mobility and hence more rapidly changing wireless channels are accounted for into the operational environment of a wireless VoIP device.

Hence it is evident that the transmission method needs to be adjusted dynamically during a VoIP call in order to guarantee adequate Quality of Service (QoS) for the users. One method to introduce adaptivity into the transmission optimization at the VoIP host is to exploit dynamically updated information about the surrounding wireless links and network elements through event triggers. For example, the trigger management mechanisms discussed in [21] would allow a VoIP host supporting multiple network access technologies to gather information on the state of its wireless or wireline interfaces so that the most suitable one can be selected for transmission. Additional information from network elements regarding e.g. congestion levels at routers along the transmission path can then be used to further optimize the transmission scheme with voice sample aggregation and ROHC.

As the WirelessMAN-OFDMA air interface used in WiMAX deployments already enables adaptation on a per user basis by allowing multiple modulation schemes and coding rates to be used inside a single DL or UL frame, further optimization at the application layer in the form of aggregation and header compression would be a natural extension to the PHY and MAC layer methods increasing the overall performance of the system. As the experimental results presented in Section IV clearly show, in good radio channel conditions, the achievable gains in link capacity are significant. By relying on the extra information on the network conditions provided by the event triggers, the VoIP host could make a justified decision to aggregate several voice samples into a single VoIP packet if the associated excess delay is not detrimental to the overall quality of the call. In addition, when the Channel Quality Indication (CQI) information gathered from the MS indicates that the wireless channel quality is sufficient, ROHC can be used to further increase the capacity without fear of constant retransmissions.

VI. Conclusion and Future Work

We measured the effective capacity of a WirelessMAN-OFDMA air interface in a state of the art WiMAX testbed compliant with IEEE 802.16 standards [5] and [6]. Although the theoretically-established potential performance of WiMAX is quite high, implementation design, configuration, and deployment practice play a crucial role in real-world performance. Using a WiMAX testbed operating at the 3.5 GHz frequency band with 5 MHz bandwidth we found that 18 simultaneous bidirectional emulated VoIP flows can be sustained without severe quality impairments. This yields a goodput of approximately 280 kb/s, which is only 5.5% and 18.3% of the maximum attainable throughput of DL and UL, respectively. By aggregating three voice samples into a single VoIP packet and employing ROHC, the number of sustained VoIP flows nearly triples.

Our empirical study highlights that the traffic mix plays an important role which cannot be overseen in future mobile WiMAX deployments. We found that the advertised effective capacity can only be attained with flows using large packets. When large amounts of flows with small packets dominate the traffic mix, our WiMAX testbed can only deliver a small fraction of its maximum capacity. We also evaluated the impact of ROHC and found that it contributes significantly less gains than VoIP aggregation. Still, ROHC may further enhance mobile WiMAX performance. Thus, dynamic application-layer adaptation according to the prevailing channel and network conditions when choosing VoIP packet sizes, as well as the possible inclusion of ROHC, is a promising concept.

The potential benefits through simple application-layer adaptation and its integration with our mobility management triggering framework are high on our future work agenda. Furthermore, as support for multiple wireless technologies is an attractive feature in today's heterogeneous network infrastructures, a solution based on the recently published IEEE 802.21 Media Independent Handover (MIH) standard is under investigation. We recently introduced an IEEE 802.21 prototype, building upon MIH services beyond mere handover preparation functionalities [22]. We plan to continue enhancing the prototype towards a more general framework in which WiMAX networks could also be included.

We are also looking forward to link capacity and VoIP performance measurements in mobile scenarios with varying traffic mixes, which were out of the scope of this study. The capacity gains from the use of Multiple-Input, Multiple-Output (MIMO) antennas for mobile WiMAX and VoIP aggregation are also to be tested in practice.

References

[1] IEEE 802.16 WG, *IEEE Standard for Local and Metropolitan Area Networks - Part 16: Air Interface for Broadband Wireless Access Systems*. IEEE P802.16-2009, May 2009.

[2] ——, *IEEE 802.16m System Description Document [Draft], Work in Progress*. IEEE 802.16m-08/003r7, February 2009.

[3] ITU-R, *Requirements Related to Technical Performance for IMT-Advanced Radio Interface(s)*. ITU-R Report M.2134, 2008.

[4] R. B. Marks, J. M. Costa, and B. G. Kiernan, "The Evolution of WirelessMAN," *IEEE Microwave Magazine*, vol. 9, no. 4, pp. 72–79, 2008.

[5] IEEE 802.16 WG, *IEEE Standard for Local and Metropolitan Area Networks - Part 16: Air Interface for Fixed Broadband Wireless Access Systems*. IEEE Std. 802.16-2004, October 2004.

[6] ——, *IEEE Standard for Local and Metropolitan Area Networks - Part 16: Air Interface for Fixed Broadband Wireless Access Systems. Amendment 2: Physical and Medium Access Control Layer for Combined Fixed and Mobile Operation in Licensed Bands*. IEEE Std. 802.16e-2005, December 2005.

[7] D. Kim, H. Cai, M. Na, and S. Choi, "Performance Measurement over Mobile WiMAX/IEEE 802.16e Network," in *Proc. WoWMoM'08*, Newport Beach, CA, USA, June 2008.

[8] J.-W. So, "A Downlink Performance Analysis of VoIP Services Over an IEEE 802.16e OFDMA System," *IEEE Communication Letters*, vol. 11, no. 2, pp. 155–157, 2007.

[9] K. Pentikousis, E. Piri, J. Pinola, F. Fitzek, T. Nissilä, and I. Harjula, "Empirical Evaluation of VoIP Aggregation over a Fixed WiMAX Testbed," in *Proc. TRIDENTCOM'08*, Innsbruck, Austria, March 2008.

[10] E. Piri, J. Pinola, F. Fitzek, and K. Pentikousis, "ROHC and Aggregated VoIP over Fixed WiMAX: An Empirical Evaluation," in *Proc. ISCC'08*, Marrakech, Morocco, July 2008.

[11] K. Correll, N. Barendt, and M. Branicky, "Design Considerations for Software Only Implementations of the IEEE 1588 Precision Time Protocol," in *Proc. Conference on IEEE 1588*, Winterthur, Switzerland, October 2005.

[12] International Electrotechnical Commission, *Precision Clock Synchronization Protocol for Networked Measurement and Control Systems*. Reference numbers: IEC 61588:2004(E), IEEE 1588-2002(E), September 2004.

[13] K. Pentikousis, J. Pinola, E. Piri, and F. Fitzek, "An Experimental Investigaion of VoIP and Video Streaming over Fixed WiMAX," in *Proc. WiOpt'08*, Berlin, Germany, March 2008.

[14] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," IETF RFC 3550, July 2003.

[15] C. Bormann, C. Burmeister, M. Degermark, H. Fukushima, H. Hannu, L.-E. Jonsson, R. Hakenberg, T. Koren, K. Le, Z. Liu, A. Martensson, A. Miyazaki, K. Svanbro, T. Wiebke, T. Yoshimura, and H. Zheng, "RObust Header Compression (ROHC): Framework and Four Profiles: RTP, UDP, ESP, and Uncompressed," IETF RFC 3095, July 2001.

[16] ITU-T, *G.729 Based Embedded Variable Bit-rate Coder: An 8-32 kbit/s Scalable Wideband Coder Bitstream Interoperable with G.729*. ITU-T Recommendation G.729.1, 2006.

[17] ——, *Coding of Speech at 8 kbit/s Using Conjugate-structure Algebraic-code-excited Linear Prediction (CS-ACELP)*. ITU-T Recommendation G.729, 1996.

[18] A. Sollaud, "RTP Payload Format for the G.729.1 Audio Codec," IETF RFC 4749, October 2006.

[19] B. A. Mah, *pchar: A Tool for Measuring Internet Path Characteristics*, Available: http://www.kitchenlab.org/www/bmah/Software/pchar/.

[20] A. Raake, *Speech Quality of VoIP : Assessment and Prediction*. John Wiley & Sons Ltd., 2006.

[21] J. Mäkelä and K. Pentikousis, "Trigger Management Mechanisms," in *Proc. ISWPC'07*, San Juan, Puerto Rico, February 2007, pp. 378–383.

[22] E. Piri and K. Pentikousis, "Towards a GNU/Linux IEEE 802.21 Implementation," in *Proc. ICC'09*, Dresden, Germany, June 2009.