A Measurement Study of VoIP over WiBro Networks

Hua Cai, Youngkyu Choi, Dongmyoung Kim, and Sunghyun Choi
School of Electrical Engineering and INMC
Seoul National University, Seoul, Korea
Email: {chai, ykchoi, dmkim}@mwnl.snu.ac.kr, schoi@snu.ac.kr

Abstract—WiBro (Wireless Broadband) network, which incorporates the IEEE 802.16e standards for providing portable Internet service, is being deployed in South Korea. WiBro network has strengths in providing better mobility support compared with WLANs (Wireless Local Area Networks) and a higher transmission rate than conventional circuit-based cellular networks. With these advantages, WiBro networks are expected to encompass the multimedia applications. Among various types of multimedia applications, especially, Voice over Internet Protocol (VoIP) is regarded as the most prosperous application to WiBro network since the demands for voice communication continue to grasp the significant position in the market. To the best knowledge of the authors, however, there has not been a work evaluating the performance of VoIP over WiBro network via measurement in the literature. In this paper, we report the result of VoIP measurement conducted in various network scenarios over commercial WiBro networks. According to our experimental results, our major conclusion is that WiBro networks can provide VoIP service with desirable quality. Also, we find that jitter loss can be a potential bottleneck performance when a VoIP session is established between WiBro users.

Index Terms—WiBro, VoIP, measurement

I. INTRODUCTION

WiBro (Wireless broadband) network, which implements the specifications defined by the IEEE 802.16e, provides high data rate IP (Internet Protocol)-based data services in a mobile environment [1]. With the explosive demands for multimedia applications, most researchers agree on that emerging wireless systems including WiBro should be able to support the quality-of-service (QoS) of various types of multimedia applications. Especially, Voice over Internet Protocol (VoIP) is regarded as the most prosperous application to WiBro network since the demands for voice communication will continue to grasp the significant position in the market.

In order to support VoIP service, WiBro system was designed in consideration of some elaborate mechanisms for VoIP at the time of technical specification [1]. The most distinguished aspect is differentiated policy for uplink resource allocation. In WiBro system, where the centralized network entity called Radio Access Station (RAS) controls the whole radio usages including uplink access, five different categories are defined to differentiate the uplink bandwidth allocation: unsolicited granted service (UGS), extended real-time polling service (ertPS), real-time polling service (rtPS), non-real-time polling service (ntrtPS), and best effort (BE). When a new connection is established, a subscriber station (SS) can specify a specific bandwidth allocation policy with associated QoS parameters such as the minimum reserved traffic rate and the maximum tolerable latency. Among those five categories, we need to look at both UGS and ertPS with respect to VoIP. Under UGS-type uplink scheduling, the bandwidth for sending a fixed size of data is allocated periodically. Accordingly, UGS is appropriate for VoIP without silence suppression. On the other hand, when the silence suppression is applied by the voice encoder, the size of VoIP packet becomes variable depending on the voice activity. In this case, the allocation of fixed bandwidth can be quite inefficient. To resolve this problem, ertPS was originally proposed by [2], and later accepted as a standard contribution officially. Under ertPS-type uplink scheduling, each SS informs RAS of the voice activity via setting one bit in the MAC header. Therefore, RAS can allocate the proper size of bandwidth without extravagancy. Similarly, downlink resource allocation policy can help the QoS of VoIP satisfied, but we do not discuss specific algorithm here since many delay-constrained scheduling algorithms proposed in the literature so far can be employed without difficulty.

In spite of these efforts, many people still have concerns about how well WiBro networks can provide the voice communication compared with conventional wireless cellular networks. Indeed, the increase of spectral efficiency, which is one of major innovations compared with circuit-based wireless systems, comes from channel state-aware scheduling over shared medium to achieve multi-user diversity. This kind of brand-new technique may deteriorate delay performance, which greatly affects the QoS of voice communication. For this reason, it is very important to evaluate and understand how good or bad state-of-the-art WiBro network can support VoIP service because the precise diagnosis can shed light on the right direction for advanced research. In this paper, we report the result of VoIP measurement conducted in various network scenarios over commercial WiBro networks.

The rest of this paper is organized as follows: In Section II, we discuss some related work. In Section III, we describes the scenarios and configurations for measurement study, and then reports the measurement results with discussion. Finally, we conclude this paper with a plan of the future work in Section IV.

II. RELATED WORK

Nowadays, emerging wireless packet data systems, e.g., HSDPA and IEEE 802.16e/WiBro, are being commercially deployed in many countries. While many of researchers have
studied those systems intensively, most efforts are based on either computer simulations or numerical analysis. For example, the throughput and delay performance of WiBro and HSDPA are compared based on computer simulations in [3]. However, both simulations and analyses have inherent limitations since they are developed from numerical models and assumptions for realities. In this context, the performance evaluation based on measurement in real systems is very meaningful since we can obtain the realistic view on the system.

Recently, there have been some efforts to evaluate the emerging wireless packet data networks based on measurement in live operational networks. In [4], the authors analyze the goodput and delay of HSDPA systems deployed in Finland. The performance metrics measured in HSDPA are compared with those of W-CDMA system, which is a precedent version of HSDPA. The delay and jitter characteristics of HSDPA are analyzed more deeply in [5]. Especially, the measurement paper for the IEEE 802.16e/WiBro system is very rare in the literature. In [6], the authors measure the traffic over WiBro systems deployed in Seoul, Korea, and then analyze the goodput and delay performance. In addition, the performance in a multi-user environment is also investigated. However, the QoS level for a specific service type is not exactly investigated.

In this paper, we just focus on the evaluation of VoIP over WiBro networks. More specifically, the QoS of VoIP service is assessed in terms of R-factor, which can reflect the impact of various impairments (e.g., delay, loss, and codec type) altogether. Then, we analyze the performance of the metric corresponding to each individual impairment. To our best knowledge, our paper is the first one, which addresses the live performance of VoIP service in the commercial IEEE 802.16e/WiBro networks.

III. MEASUREMENT STUDY OF VOIP PERFORMANCE

Before reporting our measurement results, we briefly overview our measurement environments: We carried out our measurements in a commercial WiBro network, which is deployed within Seoul National University (SNU) in Seoul, Korea. The RAS is located about 150 meters away from our measurement spot, i.e., the 4th floor at building 132 in SNU. For the consistency of measurement, we conducted the test at a specific time of the day, i.e., from mid-night to early morning, when the impact of high load and interference is thought to be minimized.

In order to evaluate the performance of VoIP, we rely on two different tools: the first one is a commercial IP performance testing toolkit IxChariot [7], which generates emulated VoIP packets according to a specific voice codec and measures the performance between a sender and destination. The second one is a popular VoIP application, Skype [8], which enables us to assess the subjective quality via real conversation and also shows some meaningful statistics such as round-trip time (RTT) and packet loss count via a built-in menu called “See the technical call information.”

For WiBro host machines, we use two laptops, i.e., IBM X31 and X60, which employ 1.6 GHz CPU and 512 Mb RAM, and connect with WiBro access network via Samsung SPH-H1100 PCMCIA-type WiBro card. To resolve the received signal strength from RAS, we use a rough measure (represented by five discrete levels as in usual cell-phone) provided by the (application-level) connection manager program. In Sections III-A and III-B, we report and discuss the results measured via IxChariot and Skype, respectively.

A. Measurement via IxChariot

We classify the measurement scenarios into two categories according to the type of underlying networks between conversational participants: 1) WiBro SS to wired host; and 2) WiBro SS to WiBro SS. The output performance metric of IxChariot is R-factor [9], [10], which can represent the QoS for VoIP by integrating the impact of one-way delay and packet loss, simultaneously. The score of R-factor ranging from 0 to 100 corresponds to a well-known subjective VoIP quality measure, called mean opinion score (MOS), as shown in Table I. The voice codec emulated by IxChariot is G. 711 with the option for packet loss concealment (PLC) and without silence suppression. Accordingly, the payload of VoIP packet is generated with the constant rate of 64 Kbps.

1) WiBro SS to wired host: In this scenario, we configure network topology as shown in Fig. 1: a laptop, to which the IP address of 125.149.68.38 is dynamically allocated by WiBro network, establishes a VoIP session with a desktop host connected by Ethernet to the Internet. Here, the one-way delay can be divided into two components: 1) communication delay (the total time a packet sent from the physical interface of either terminal takes to reach the partner’s physical interface); and 2) processing delay in voice codec plus jitter-buffer delay. The second component delay seems to be different depending on the codec type used, but looks likely to be constant for a given codec type.

<table>
<thead>
<tr>
<th>R-factor</th>
<th>MOS</th>
<th>Remarks</th>
</tr>
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<tbody>
<tr>
<td>90</td>
<td>4.34</td>
<td>Very satisfied</td>
</tr>
<tr>
<td>80</td>
<td>4.03</td>
<td>Satisfied</td>
</tr>
<tr>
<td>70</td>
<td>3.6</td>
<td>Some users dissatisfied</td>
</tr>
<tr>
<td>60</td>
<td>3.1</td>
<td>Many users dissatisfied</td>
</tr>
<tr>
<td>50</td>
<td>2.58</td>
<td>Nearly all users dissatisfied</td>
</tr>
</tbody>
</table>
Since IxChariot gives both one-way delay and communication delay as output statistics, we can interpret their difference as the codec plus jitter-buffer delay for the G. 711 voice codec in consideration. Under this motivation, we first measured the delay statistics without imposing any artificial delay. The statistics was obtained by averaging a sufficient number of measurement runs and each run spans 150 seconds. Throughout the measurement, the received signal strength amounted to 5/5 (very good). Fig. 2 shows that the difference between one-way delay and communication delay is time-invariant and fixed at 41 msec. Therefore, we interpret this value as the codec plus jitter-buffer delay for the rest of this paper.

Next, in order to understand the impact of one-way delay on the QoS of VoIP, we impose artificial delay additionally in a unit of 50 msec at the laptop. Fig. 3 shows the R-factor for a given artificial delay with a 95% confidence interval. With respect to R-factor of 80 (corresponding to “most users are satisfied” in Table I), we observe that the delay margin as long as 150 msec is conserved. Accordingly, even if the talk partner is located geographically apart (which means that the one-way delay is quite long), WiBro network will be able to support VoIP communication in most cases since the one-way delay is highly likely to be less than 150 msec in the same continent [11].

Also, when the channel quality becomes worse (4/5 corresponding “good”), we see that the R-factor achieved decreases in moderate range of artificial delay (from 100 to 200 msec). This can be explained as follows: we already know that error control schemes like automatic repeat request (ARQ) increase the mean and variance of the packet delivery delay over WiBro wireless link as the packet error events occur more frequently due to worse channel quality. Accordingly, the one-way delay obtained at the channel quality of 4/5 is a little bit longer than at the channel quality of 5/5. On the other hand, the E-model in [9] tells us that the delay impairment increases abruptly when the one-way delay starts to exceed a certain threshold value. Therefore, the impact of worse channel quality looks distinct only near the threshold value.

2) WiBro SS to WiBro SS: In this scenario, we replace an Ethernet host with another WiBro SS as shown in Fig. 4. Compared with the previous scenario, now VoIP session has the communication delay composed of uplink plus downlink delays over WiBro network. All the measurement results are obtained when the channel quality is very good (i.e., 5/5). In Table II, we summarize the notable figure of merits in comparison with the result obtained in Section III-A.1.

First, we see that the one-way delay increases by as much as 25 msec. From the differences between two network topologies shown in Figs. 1 and 4, the increase of delay should be attributed to the WiBro downlink path.

Second, we notice that the jitter increases remarkably, which means that the VoIP packet experiences more delay variations. For this reason, 1.7% VoIP packets was lost at the de-jitter buffer, which smoothes out the delay variations of the received voice frame to play out the decoded voice in a synchronous stream. The packet losses at the de-jitter buffer are caused by either buffer overrun or under-run. Under-run occurs when a certain number of consecutive packets arrive at the receiver too late, and hence the jitter buffer becomes empty. In this case, the listener feels disruption in the middle of partner’s talk and the packets arriving too late are dropped since they are useless. Overrun happens when the jitter buffer is already full at the time of packet arrival. Also, in this case, the packet is dropped due to buffer overflow. Both buffer overrun and under-run are accelerated as jitter performance becomes worse.

In our measurement, we set the size of jitter buffer to two
IP datagrams, which yields a sufficiently small jitter value between a WiBro SS and wired host. Accordingly, the jitter could be a bottleneck performance for overall QoS of VoIP, and hence the size of jitter buffer for the VoIP session between WiBro SSs should be set preferably to a larger value compared with the case between WiBro SS and an Ethernet host. At this time, however, we should be also careful for the trade-off that the increase of buffer size yields a longer one-way delay. For this reason, even though the achieved QoS of VoIP reported in Table II still looks somehow good enough to enjoy voice communication, we conduct additional measurements to understand the relationships among the jitter buffer size and three important performance measures, i.e., packet losses, one-way delay, and R-factor.

**Impact of jitter buffer size**: For the jitter buffer size ranging from 2 to 8 datagrams, we record packet losses, one-way delay, and the average R-factor. From Fig. 5(a), as the size of jitter buffer increases, we observe that the number of packet losses is sharply reduced. As our expectation, this tells us that increasing the size of jitter buffer is an effective way to reduce the packet loss due to jitter. When the buffer size is 2, more than 3% packets get lost, but packet losses almost disappear when the size of jitter buffer is adjusted to 5.

Fig. 5(a) also shows the one-way delay with varying the size of jitter buffer. The one-way delay increases with jitter buffer size because the received packets should stay more at the buffer before being decoded. Interestingly, the increase in one-way delay does not look constant, but shows regular pattern: the one-way delay increases either by 10 msec or by 20 msec. Recalling that the inter-packet arrival time is 20 msec, the amount of delay increase corresponds to the half or one inter-packet arrival time, respectively.

Due to this trade-off relationship between packet losses and one-way delay, the proper value for jitter buffer size should be optimized with respect to the R-factor, which integrates the impact of delay and packet loss. Fig. 5(b) shows that R-factor increases up to the jitter buffer size of five due to smaller packet losses and it starts to decrease gradually with further increase of jitter buffer due to excessively degraded one-way delay. For this reason, R-factor achieves the peak value when the jitter buffer size is 5. Even though the effect from the optimization of jitter buffer size does not look significant, it is important to understand the trade-off happening when we adjust the size of jitter buffer to mitigate the impact of jitter.

**B. Measurement via Skype**

In order to evaluate the subjective QoS of VoIP, we establish real VoIP sessions using a popular VoIP application, Skype, and make conversations at various channel qualities (e.g., 3/5, 4/5). For a given channel quality, we repeat talking via VoIP ten times and each conversation lasts for more than one minute. The underlying network topology is WiBro SS to WiBro SS shown in Fig. 4. From the perspective of the subjective quality of conversation, the quality of VoIP was pretty good even at the channel quality of 3/5 since we could hear the voice clearly without any delay or interruption.

Fig. 6 shows the screen-shot of a built-in menu called “See the technical call information,” which provides the VoIP statistics of Skype. At the end of each VoIP session, we record the round-trip time and summarize in Table III.

**IV. CONCLUSION**

In this paper, we evaluated the service quality of VoIP service based on the measurements in a commercial WiBro.
system. We measured delay and packet loss rate, and translated it to R-factor which is a QoS metric for VoIP service. At the all environments in this paper, the measured R-factor exceeded 85 which indicates very good quality of VoIP service. Up to 150 ms of additional delay in the Internet was also acceptable. From the results, we conclude that the current WiBro system can support VoIP well in the general environments. Furthermore, we showed that the performance can be enhanced by optimizing the size of jitter buffer. Finally, we verified our conclusion by analyzing the performance of a commercial VoIP application over WiBro.

REFERENCES


