An Empirical Study on the Quality Assessment of the VoIP Service over Wireless Mobile Networks

Kwang-Yeol Yoon and Beomjoon Kim*

Department of Electronic Engineering, Keimyung University, Daegu, 704-701, Korea
{yoonky,bkim}@kmu.ac.kr

Abstract. This paper discusses the service quality of packet-based voice service provided over wireless mobile networks such as wireless broadband (WiBro) and high speed downlink packet access (HSDPA) systems. Using measurement software, a large scale of experiment has been conducted to measure the actual quality of the voice service over both wireless mobile networks. Based on the results from the experiment, the quality of the voice service is supposed to be quite good. Through further experiment, the quality degradation over a radio channel leads to the increase in delay and the subsequent quality degradation of the voice service.

Keywords: mobile VoIP service, voice quality measurement software, E-Model, mean opinion score (MOS).

1 Introduction

The packet-based voice service such as voice over IP (VoIP) service over a wired network [1] has become popular now because of its low cost. In providing the voice service over a wireless mobile network, however, one of the challenges that should be addressed is how to support and guarantee its service quality [2], [3]. Not only the bandwidth provided by a wireless channel is small, several characteristics should be considered; e.g., the time-varying channel qualities over a wireless channel, mobility-related processes like handoff, and etc.

For the past few years, we have studied to find the answers for the following two questions:

- What is the current service quality of voice service over wireless mobile networks?
- What level of radio channel quality provokes the degradation of the service quality of the voice service over wireless mobile networks?

And the answers are presented in the remainder of this paper.

* Correspoding author.

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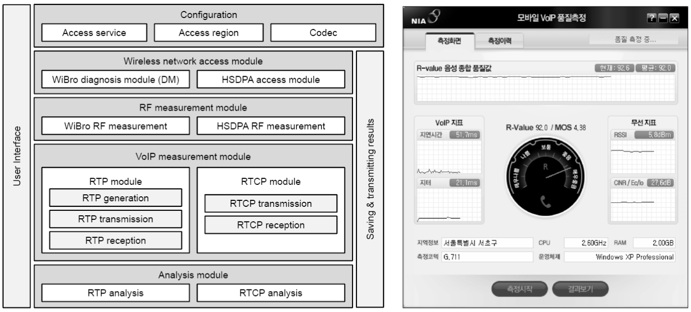
2 The Quality Measurement Software

In order to measure the voice quality in this study, we use the software developed in our previous research [4]. Fig. 1-(a) shows the logical architecture of the software running on the user equipment's side.



(a) the logical architecture (b) a screen shot of the output

Fig. 1. The voice quality measurement software

The software consists of the measurement function, the reporting function, and the user interface. It supports the access to two wireless mobile networks that are available now; one is wireless broadband (WiBro) that is based on IEEE 802.16 standard and well-known as WiMAX and the other one is high speed downlink packet access (HSDPA) that is a 3G mobile system.

Table 1. The selected service quality metrics

	Quality metrics	
	WiBro	HSDPA
RF quality metrics	RSSI (received signal strength indicator)	
	CINR (carrier to interference noise ratio)	Ec/Io (energy per chip over the interference noise)
Network quality metrics	Bandwidth, one-way delay, jitter, packet loss rate	
VoIP quality metrics	R-Score, MOS (mean opinion score)	

Considering the communication architecture in which voice packets are delivered, the measurement function has been implemented by the RF measurement module and the VoIP measurement module. A few quality metrics are selected in consideration of their importance and summarized in Table. 1. The RF quality metrics are measured by the RF measurement module while the network and VoIP quality metrics by the VoIP measurement module.

The wireless quality metrics are measured based on the information obtained through the standard interface and commands provided by the modem manufacturer. The RFCs are referred to for the algorithm to measure the network quality metrics;

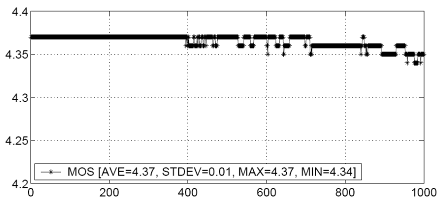
i.e., one-way delay [6], packet loss ratio [7], and jitter [8]. For the VoIP quality metrics, we follow the E-Model [5] that derives R-Score and mean opinion score (MOS) using the measured values of network quality metrics. Fig. 1-(b) shows a screen shot generated by the software with the measured values.

3 Experiments

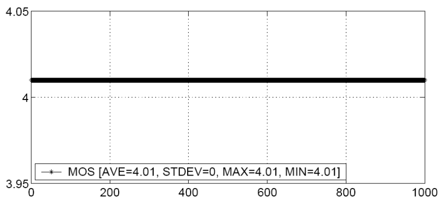
3.1 Experiment 1

In order to investigate the current status, we have conducted an experiment to measure the VoIP service quality over wireless mobile networks. The measurement software runs on the user equipment that is implemented by a laptop. The voice traffic for test generated by a server is delivered to the user equipment through actual WiBro and HSDPA mobile networks. Two codecs are considered; one is G.711 that requires 64kbps and the other one is G.729 that requires 8kbps with compression.

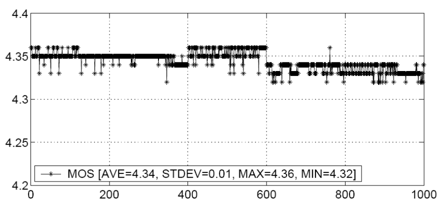
Reflecting various transmission characteristics over a wireless link, the measurement is performed under the following five conditions: indoor-stationary (E1), indoor-walk (E2), outside-stationary (E3), outside-walk (E4), and outside-moving (E5). A measurement period lasts for 300 seconds. A data sample is made from averaging 60 values measured in every 5 seconds during the measurement period. For each scenario, 200 samples are obtained and the results are shown in Fig. 2.



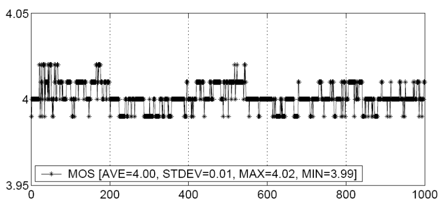
(a) WiBro (G.711)



(b) WiBro (G.729)



(c) HSDPA (G.711)



(d) HSDPA (G.729)

Fig. 2. The measured MOS (x-axis: number of sample, y-axis: the measured value)

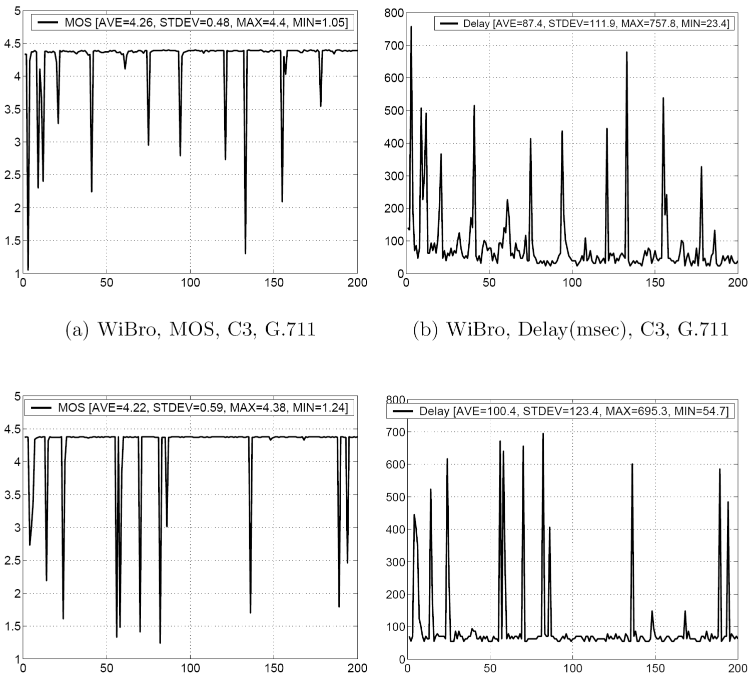
During the measurement, the bandwidth measured at the user equipment ranges from 5.93 to 7.82 for WiBro and from 2.09 to 3.18 for HSDPA in average. Considering that even G.711 that adopts no compression requires only 64 kbps, the provided bandwidth is quite sufficient to accommodate the voice traffic. Although the MOS values in four diagrams decrease a little for the samples 800 through 1000 comprising E5, the overall value shows better than 4.0 that corresponds to 'toll-quality'. Note that the highest value of MOS with G.729 is no better than G.711 because of compression.

3.2 Experiment 2

In Experiment 2, we have conducted a sort of "stress-test" to figure out the RF channel quality that provokes the voice quality degradation. For this experiment, it should be noted that a radio channel emulator is used to adjust the radio channel quality artificially. We have also determined three configurations applied to the radio channel emulator as in Table 2. For each configuration, 200 samples are obtained for WiBro and HSDPA in terms of G.711 and G.729 in respect.

Table 2. The RF channel configuration for experiment 2

	WiBro	HSDPA
Configuration 1 (C1)	RSSI -70(dBm), CINR 16(dB)	RSSI -65(dBm), Ec/Io -5 (dB)
Configuration 2 (C2)	RSSI -75(dBm), CINR 14(dB)	RSSI -65(dBm), Ec/Io -10(dB)
Configuration 3 (C3)	RSSI -80(dBm), CINR 12(dB)	RSSI -65(dBm), Ec/Io -15(dB)



(a) WiBro, MOS, C3, G.711 (b) WiBro, Delay(msec), C3, G.711

(c) HSDPA, MOS, C3, G.711 (d) HSDPA, Delay(msec), C3, G.711

Fig. 3. The measured MOS and delay for WiBro and HSDPA
(x-axis: number of sample, y-axis: the measured value)

Fig. 3 shows the measured results for WiBro and HSDPA. Since every measured value of MOS maintain above 4 for C1 and C2, we have included the results only for C3. In Fig. 3-(a), it is observed that sometimes MOS decreases very sharply and deeply. The reason for such decrease can be found in the measured delay values shown in Fig. 3-(b). It can be shown that the measured delay value increases very high at the same number of sample where MOS decreases. Almost similar results are observed for HSDPA as shown in Fig. 3-(c) and 3-(d). For G.729, the MOS values in every measurement always stay above 4, which can be explained that it requires only 8 kbps.

4 Conclusion

In this paper, we have investigated the service quality of VoIP service over wireless mobile networks. Based on the measurement results from our experiments, it can be concluded that WiBro and HSDPA networks provide sufficient bandwidth so that the voice quality is quite good and the degradation over a radio channel leads to the increase in delay and the subsequent degradation of the voice quality.

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