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Mobile-Taiwan experience in voice over IP-worldwide interoperability for microwave access trial

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Abstract: Considering voice as a dominant telecommunication service, the performance of Voice over IP (VoIP) plays a critical role in deployment of worldwide interoperability for microwave access (WiMAX) technology providing all-IP network services. To that effect, in this study, the authors investigate the performance of a WiMAX-based VoIP established under the mobile Taiwan (M-Taiwan) field-trial funded program. To achieve the objectives of the trial the measurement results expressed in the form of mean opinion score (MOS), packet loss, packet delay and jitters. For the worst-case scenario, the tests were conducted under a stringent condition of both communicating devices, wirelessly connected to the same WiMAX base station under a heavy background traffic and interference, were experiencing simultaneous handovers during the communication. Upon their analysis, the field measurements confirm an excellent performance when both communicating devices kept stationary and show an acceptable quality for the service when both communicating devices are on the move at a speed of 50 km/h.

1 Introduction

Taiwan's Wi-Fi industry accounts for more than 90% of the global market share. In its quest to identify the next-generation products, the Taiwan government has chosen worldwide interoperability for microwave access (WiMAX) [1] as one of the major directions for Taiwan's wireless industry, and has established the Mobile Taiwan (M-Taiwan) Program as the blueprint for an island-wide WiMAX environment. M-Taiwan aims at developing chip sets and base stations (BS). For example, WiMAX chip sets have been developed by Mediatek, and the BSs have been developed by T-Com and ZyXEL. Furthermore, by creating its own WiMAX ecosystem, Taiwan offers not only manufacturing capabilities, but also an entire service and application test-bed for mobile services, mobile learning and mobile life. Since 2006, 18 large-scale WiMAX service trials have been deployed in Taiwan [2, 3].

In M-Taiwan, the voice over IP (VoIP) service is considered as an enabling technology integrating broadband data applications with the voice. In particular, IP multimedia core network subsystem (IMS) [4] is utilised for voice and data integration. Under the support of the M-Taiwan Program, this paper investigates the VoIP performance for a WiMAX deployment in Taipei City. In this paper, we elaborate on the VoIP experimental environment, describe the output measures and demonstrate the VoIP performance with and without mobility. The remainder of this paper is organised as follows. Sections 2 and 3 provide a brief overview of VoIP service and WiMAX system. The general configuration set-up for the experimental field tests explained in Section 4 followed by the service performance measurement system in Section 5 and detailed results in Section 6.

2 VoIP overview

With the explosive growth of the Internet subscriber population, VoIP has become the most promising low-cost option for voice communication over the IP network. In the M-Taiwan Program, VoIP is implemented by using the session initiation protocol (SIP) [5] and real-time transport protocol (RTP) [6].

2.1 SIP and RTP

IETF RFC 3261 defines SIP for Internet telephony [5]. As an application-layer signalling protocol over the IP network, SIP is designed for creating, modifying, and terminating multimedia sessions or calls. A SIP customer premise equipment (CPE) is installed with a user agent. The user agent contains both a User Agent Client (UAC) and a User Agent Server (UAS). The UAC (or calling user agent) is responsible for issuing SIP requests, and the UAS (or called user agent) receives the SIP requests and responds.

The SIP message specifies the RTP [6], which delivers the data in the multimedia sessions. Implemented on top of UDP, RTP detects packet loss and ensures an ordered delivery. The RTP packet also indicates the packet sampling time from the source media stream. The destination application can use this time stamp to calculate delay and jitter to provide the QoS feedback.

SIP conjuncts with protocols such as session description protocol (SDP) [7] to describe the multimedia information. It conveys sufficient information to enable applications to join a session. During the session initiation, SDP describes the media type, media protocol and codec number supported by the session endpoints to announce the endpoints capabilities. SDP provides the RTP information such as the network address and the transport port number of the RTP connection. Details of SIP and RTP can be found in [4].

2.2 E-model

The quality of a communication service is traditionally based on subjective perception and typically measured by the mean opinion score (MOS), which considers the effects of equipment and impairment factors to subjectively quantify the perceived quality of a transmission such as voice based on typical users' perceptions [8]. The MOS value ranges are quantised to five levels, from 1 to 5, where 1 is unacceptably bad, 2 is poor, 3 is fair, 4 is good and 5 is excellent. The ITU-T G.107, however, defines an E-model, which provides a computational model for rating the end-to-end transmission performance for the VoIP service [8]. The E-model considers different kinds of transmission impairments add on linearly to the scale of the rating factor R. The model then converts the value of R into a MOS scale that quantifies an overall conversational quality.

The rating factor R is then expressed as follows

$$R = R_o - I_s - I_d - I_{e\text{-eff}} + A \tag{1}$$

In the right-hand side of (1), these factors are described as follows:

R_o: the basic signal-to-noise ratio includes the noise sources such as circuit noise and room background noise.

I_s: the simultaneous impairment factor combines the impairments that occur simultaneously with the voice signal. These impairments include the quality degradation caused by the overall loudness, non-optimum sidetone and quantising distortion.

I_d: the delay impairment factor represents the impairments because of delay in arrival of the voice signal.

$I_{e\text{-eff}}$: the effective equipment impairment factor represents impairments caused by low bit rate CODEC and the impairments because of random packet loss.

A: the advantage factor allows for compensation of impairment factors when there are other advantages of access to the user. ITU-T G.107 suggests the default value 0 for A.

The rating factor R is then converted into an estimated MOS value as follows

$$\begin{cases} \text{For } R < 0: \quad \text{MOS} = 1 \\ \text{For } 0 < R < 100: \quad \text{MOS} = 1 + 0.035R + R \times (R - 60) \\ \quad \times (100 - R) \times 7 \times 10^{-6} \\ \text{For } R > 100: \quad \text{MOS} = 4.5 \end{cases} \tag{2}$$

Therefore the estimated MOS values range from 1 to 4.5. The relation (2) between the estimated MOS value and the rating factor R is illustrated in Fig. 1.

3 WiMAX overview

Following the success of the Internet technology, broadband data communication services have been provisioned to the expert communities for decades, which for the wired and fibre connections have been achieved with the turn of the

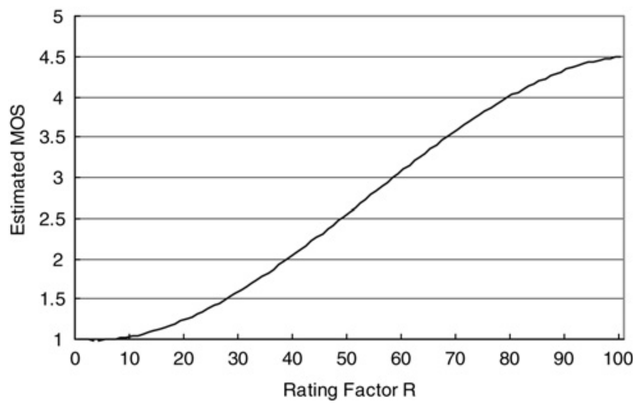

Figure 1 *Estimated MOS value as a function of rating factor R*

century. For wireless it is due any time within the next decade where superior mass production of quality wireless components to extend the frequency range and overcome shadowing and multipath fading issues using super sensitive receivers [9]. Now, with the industry capable of providing the WiMAX technology for superiority of virtually nil infrastructure costs, we are able to offer a data-enabled very low cost wireless metropolitan area network (WMAN) style wireless broadband access (WBA) solutions, which in long run may overshadow competitive solutions [10] because of the fact that WiMAX is able to provide broadband wireless access with wide service coverage, high data throughput, high mobility and greater service flexibility [5, 11, 12]. Fig. 2 shows a simplified WiMAX network architecture, which consists of the access service networks (ASNs; see Fig. 2*a*) and the connectivity service networks (CSNs; see Fig. 2*b*). An ASN provides radio access (such as radio resource management, paging and location management) to the WiMAX mobile station (MS; Fig. 2*e*). The ASN comprises ASN gateways (ASN-GWs; see Fig. 2*c*) and WiMAX BSs (see Fig. 2*d*). Every ASN-GW connects to several BSs. The ASN-GWs are also connected to each other to coordinate MS mobility. A CSN consists of network nodes such as the mobile IP (MIP) home agent (HA; see Fig. 2*f*) [8], the authentication authorisation, and accounting (AAA) server (see Fig. 2*g*) and the dynamic host configuration protocol (DHCP) server (see Fig. 2*h*). The CSN provides IP connectivity (such as Internet access and IP address allocation) to a WiMAX MS and interworks with the ASNs to support capabilities such as AAA and mobility management. Before an MS is allowed to access WiMAX services, it must be authenticated by the ASN-GW (which serves as the authenticator) and the AAA server in the CSN. Details of WiMAX technology can be found in [3, 9].

4 VoIP experimental environment

The main bulk of this trial service performance measurement has been conducted during 2007–2008 in the Taipai area under various communication conditions. Based on the abstract network shown in Fig. 2, Fig. 3 illustrates the network architecture for one of the WiMAX deployments

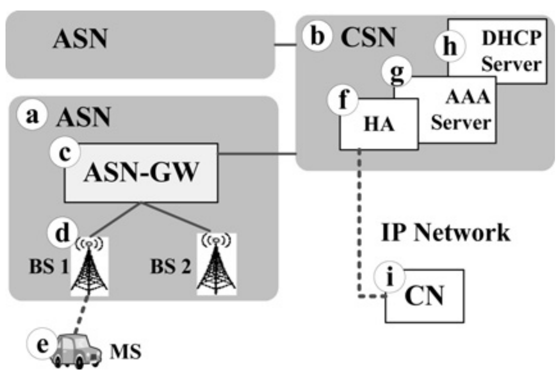

Figure 2 *Simplified WiMAX network architecture*

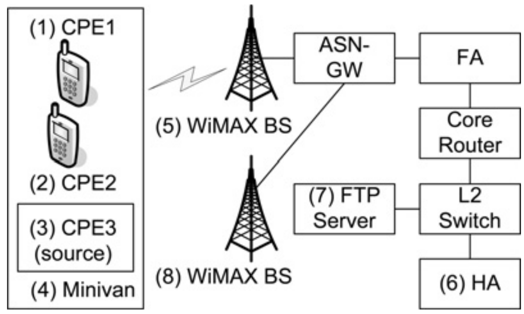

Figure 3 *M-Taiwan VoIP experimental environment*

in the M-Taiwan Program. Based on mobile WiMAX (IEEE 802.16e-2005) technology [1], more than 52 WiMAX BSs have been deployed. The WiMAX ASN-GW (a Foundry's Netlron XMR400 plus Motorola's CAP Controller) is located in Taipei County. The distances between the BSs to be tested in our study and the ASN-GW range from 18.5 to 21 km. Every BS is connected to the ASN-GW through a 50 Mbps optical fibre link. The ASN-GW connects to the foreign agent (FA; which is a Redback's SmartEdge 400) through gigabit ethernet (GE). The FA connects to a core router (Juniper's M120) through another GE. The core router connects to an L2 switch (Cisco's Catalyst 3560E) through GE. The L2 switch connects to the HA (a Starent's ST-16 Intelligent Mobile GW) through GE, and connects to an FTP server through a 10/100 M fast ethernet (FE). In the above configuration, backup for ASN-GW controller, FA and core router are also deployed to support reliability and availability.

In this experimental environment, the WiMAX MSs are installed SIP call agents, and serve as SIP CPEs. The VoIP calls are generated and measured between WiMAX CPE1 (Fig. 3 (1)) and WiMAX CPE2 (Fig. 3 (2)). Our experiments also include the background data traffic, which is generated from WiMAX CPE3 (Fig. 3 (3)) to the FTP server (Fig. 3 (7)). These three CPEs are notebooks connected to Quanta/Beceem's WiMAX (wave 2) USB dongles, and are all located in a minivan (see Fig. 3 (4)). As illustrated in Fig. 4*a*, it is clear that a one-way VoIP link between CPE1 and CPE2 consists of 12 hops (CPE1 ⟷ BS ⟷ ASN-GW ⟷ FA ⟷ core router ⟷ L2 switch ⟷ HA ⟷ L2 switch ⟷ core router ⟷ FA ⟷ ASN-GW ⟷ BS ⟷ CPE2). In Fig. 4*b*, the data path between CPE3 and the FTP server includes six hops (CPE3 ⟷ BS ⟷ ASN-GW ⟷ FA ⟷ core router ⟷ L2 switch ⟷ FTP server).

In our study, a three-sector WiMAX BS is typically installed at the roof of a building with the coverage of 1.5 km in diameter. To fully utilise existing cellular infrastructure, the WiMAX antenna may be collocated with the WCDMA antenna. The WiMAX antenna is an adaptive system with beamforming. In this WiMAX network deployment, the time division duplex (TDD) ratio for downlink and uplink can be 3-to-1 or 3-to-2. In our

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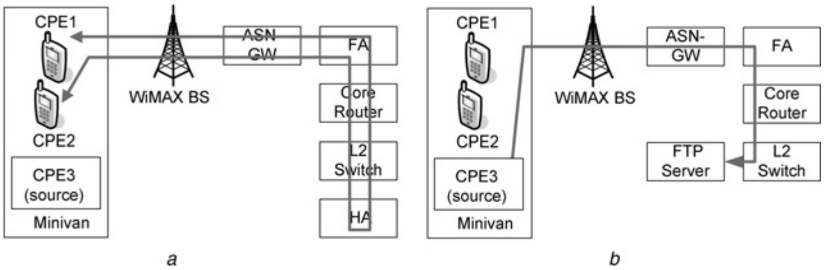

Figure 4 *Data paths in the experiments*

a VoIP path between CPE1 and CPE2
b FTP path between CPE3 and FTP server

experiments, 3-to-1 ratio is considered. The modulation schemes are 16-quadrature amplitude modulation (16QAM) 3/4, 16QAM 1/2, quadrature phase shift keying (QPSK) 3/4, QPSK 1/2 for uplink, and 64QAM 5/6, 64QAM 3/4, 64QAM 2/3, 64QAM 1/2, 16QAM 3/4, 16QAM 1/2, QPSK 3/4, QPSK 1/2 for downlink. We observed that the bandwidth performance is significantly improved by up to 100% in our measurements when the modulation scheme is enhanced from 64QAM 1/2 to 64QAM 5/6 for downlink, and from 16QAM 1/2 to 16QAM 3/4 for uplink. Through measurements of 14 experiments, the average TCP uplink transmission rate is 3.668 Mbps. Fig. 5*a* plots a typical experiment of TCP uplink transmission rate as the CPE speed changes. The sample points are measured for every 2–3 s. The figure indicates that the transmission rate drops significantly as the CPE suddenly accelerates (e.g. when the speed increases from 6 to 59 km/h).

The average downlink TCP transmission rate of the BSs is 10.01 Mbps (per sector). Fig. 5*b* plots a typical experiment of TCP downlink transmission rate as the CPE speed changes.

For a stationary CPE, the maximum and minimum uplink TCP bandwidths are 2.879 and 2.306 Mbps, respectively. The average uplink bandwidth is 2.492222 Mbps. The maximum and minimum downlink TCP bandwidths are 7.781 and 4.741 Mbps, respectively. The average downlink bandwidth is 6.881778 Mbps.

We also measure the handover delays. The average handover delays of five measurements are 67.78 ms for inter-BS handover at 30 km/h, 68.125 ms for inter-BS handover at 50 km/h, 63.5 ms intra-BS handover at 30 km/h and 65 ms for intra-BS handover at 50 km/h. Therefore as the CPE speed increases form 30 to 50 km/h, the handover delay increases by 0.51–2.3%. The inter-BS handover time is 4.8–6.7% longer than the intra-BS handover.

In the stationary tests, the distance between the CPEs and the BS is about 210 m, and the output data are measured at a BS located at the seventh floor of a building in Nei-Hu area of Taipei City. In the mobility tests, two WiMAX BSs are involved. These BSs are located near the Taipei City Hall. To produce the handover effect under the controlled condition, the minivan carrying the CPEs repeatedly drove on the roads around a square area adjacent to the City Hall (see Fig. 6). This path is covered by two WiMAX BSs and the distances between the BSs and the CPEs range from 150 m to more than 400 m.

5 VoIP experimental setup for output measurement

As described in Section 2.1, in M-Taiwan, SIP is used to control a VoIP call and RTP is used to deliver the voice data. In this paper, we focus on the RTP packet performance. The SIP call setup signalling can be found in [4], and will not be discussed in this paper.

We use the NetIQ Chariot tool [13] to measure the MOS values following the E-model described in Section 2.2. To collect the measured data, the Chariot endpoints are installed in the VoIP CPEs running on Windows XP 6.2. The Chariot Console resides in one of the CPEs.

In VoIP, CODEC are used to convert analogue voice to digital samples so that the voice information can be delivered in the IP network. The VoIP codec techniques determine the maximum MOS value. Our experiments utilise the high-quality G.711 codec that consumes larger bandwidth as compared with other CODEC. The maximum MOS value for G.711 is 4.4 (which is lower than the theoretical value 5). Although other CODEC such as G.729 (with maximum achievable MOS value 4.07) are also supported in M-Taiwan deployments, G.711 is selected for presentation in this paper because WiMAX support broadband applications, and therefore can comfortably accommodate G.711 CODEC. The default G.711 data rate is 64 kbps.

There are several techniques to improve the performance of the codecs. With silence suppression, when no one is talking during a call, VoIP data are not delivered to save the network bandwidth. In our experiments, silence suppression is

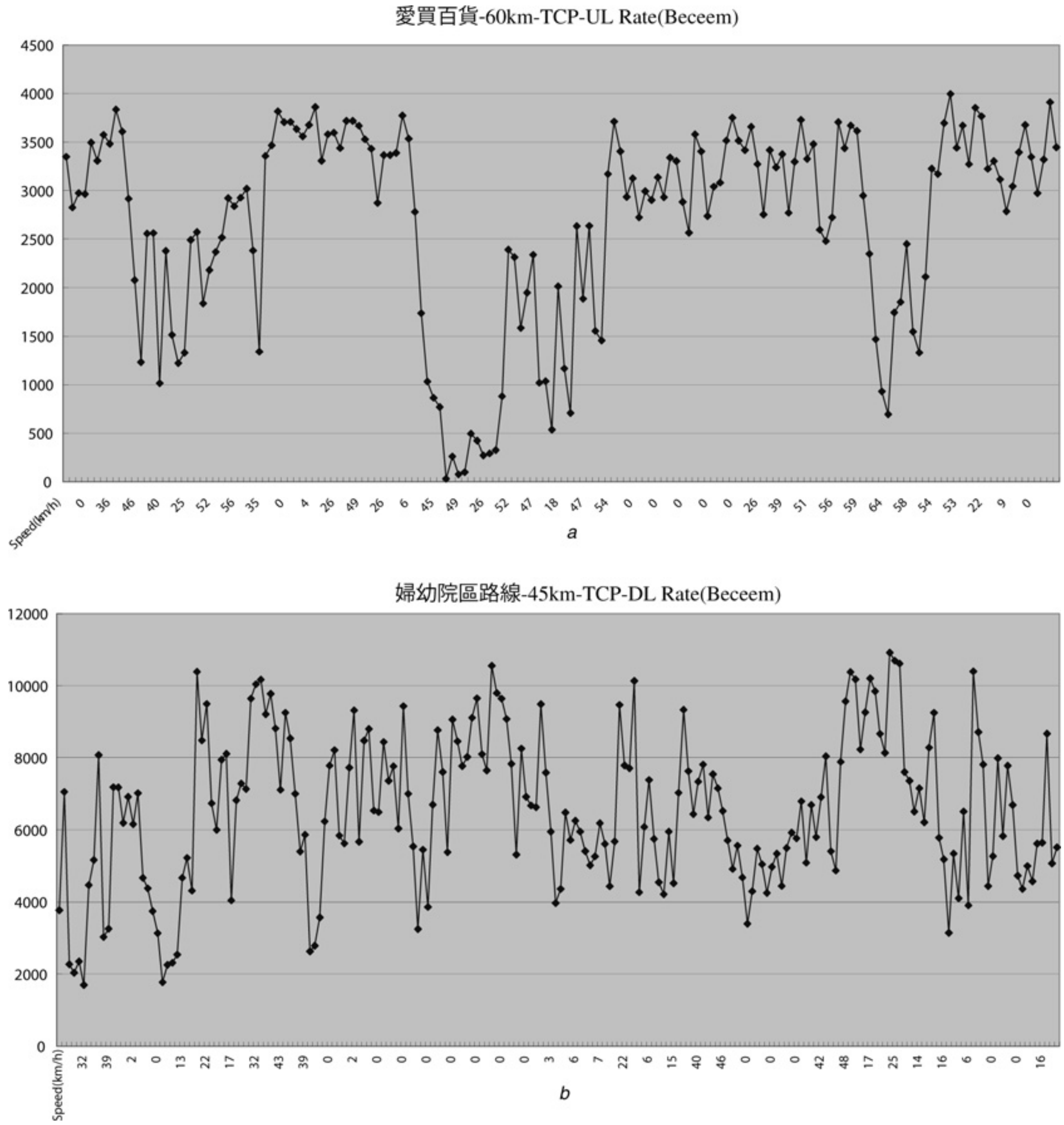

Figure 5 *Real-time measures of TCP transmission rate at various CPE speeds*
a TCP uplink transmission rate (kbps) against CPE speed (km/h)
b TCP downlink transmission rate (kbps) against CPE speed (km/h)

disabled to obtain a more intense assessment of the network. The G.711 packet loss concealment (PLC) option mitigates the VoIP data loss effects and therefore improves the MOS estimate. This option is turned off in our experiments for a stringent scenario investigation.

Besides MOS, the following output measures are also investigated in this paper:

Packet loss can severely impair call quality because voice information cannot be received by the listeners. Data loss in bursts is more serious than uniform random loss because humans perceive bursts of loss as impairments to audio quality much more than uniform random loss. Bursts of loss are often observed in radio links, and are a major measure we would like to investigate in this study. Packet loss is included in the MOS calculation (see factor $I_{\text{e-eff}}$ in Section 2.2). WiMAX Forum requires that packet loss be less than 1%.

One-way packet delay is the time period for a VoIP packet to travel from one CPE to another. Typically, the voice quality is acceptable when the voice delay is less than 150 ms. When it exceeds 200 ms, the listeners will experience the walkie-talkie effect with poor audio quality [8]. In our experiments, the one-way packet delay includes the propagation delay contributed by 10 IP hops (from the BS to the HA and back), the transport delay contributed by two WiMAX radio links, the G.711 packetisation delay and jitter buffer delay. G.711 introduces packetisation delay to convert a signal from analogue to digital. In our experiments, the packetisation delay is set to 20 ms. This delay is included in the MOS estimate (see factor I_{d} in

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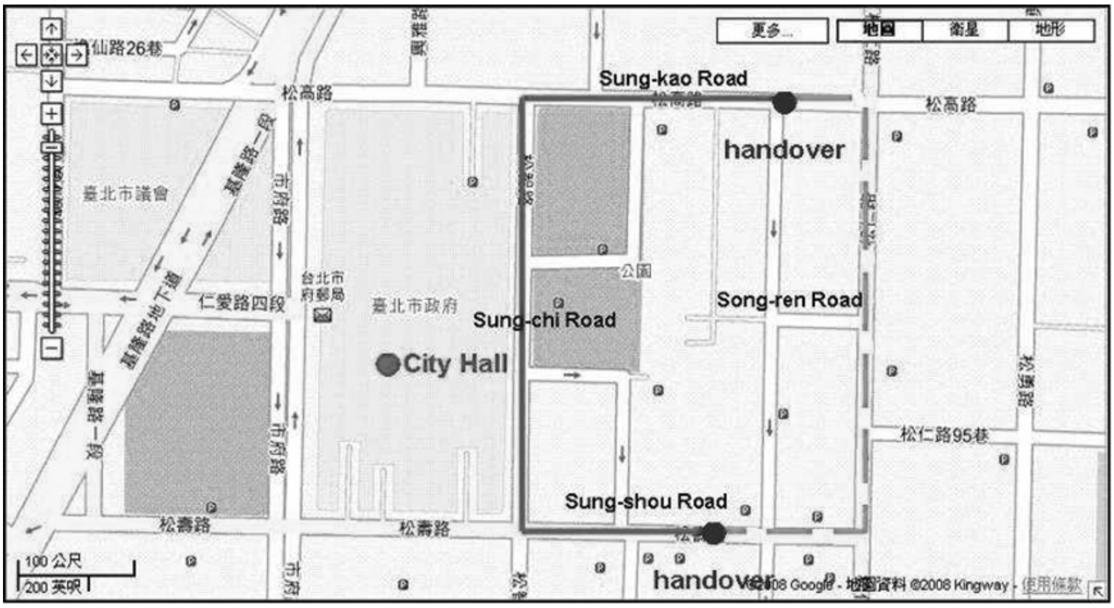

Figure 6 *Moving path for mobility tests (the solid path is covered by one base station and the dashed path is covered by another base station)*

Section 2.2). The WiMAX forum specifies the preferred packet delay to be less than 150 ms, and the limited delay to be 200 ms.

Jitters or the variation of packet inter-arrival time may create unexpected pauses between utterances, and therefore affect the intelligibility of the VoIP speech. It was reported that an average jitter exceeding 35 ms [14] or 50 ms [13] results in unacceptable QoS for VoIP. The WiMAX forum requires that jitter is less than 25 ms. In order to reduce the jitters, a buffer is used to store the incoming packets before they are played. If the jitter buffer size is too small, network jitter will result in packet loss and therefore degrade the intelligibility of the voice. If the jitter buffer size is too large, long packet delay will be experienced, which results in QoS degradation (e.g. echo level may be more easily perceptible). Our previous study indicated that buffering one packet is sufficient for WLAN if the core network delay (transport delay) is not considered [12]. Default G.711 jitter buffer delay is 40 ms (two packets) in our experiments, which is also included in the MOS estimate.

In some wireless VoIP experiments, only one call party resides in wireless network and the other call party is directly connected to Internet through wired network [11, 12]. We consider a stringent scenario where both CPEs (CPE1 and CPE2 in Fig. 3) are wirelessly connected to the same WiMAX BS, and will handover at the same time. To our knowledge, the behaviour of simultaneous handovers for both call parties is seldom reported in the literature. We also use a third CPE (CPE3 in Fig. 3) to generate the uplink background traffic. Downlink background traffic is not considered because the WiMAX uplink is the bottleneck (due to the 1-to-3 uplink-to-downlink bandwidth ratio), and our past experience indicated that the impact of WiMAX uplink background traffic is more significant than that of downlink background traffic.

Based on the configuration illustrated in Fig. 3, there are two VoIP links and one background traffic link in every experiment. The background traffic with 512 kbps, 1, 2 and 3 Mbps are considered. We note that the 3 Mbps background traffic consumes most of WiMAX uplink bandwidth. In terms of CPE mobility, we consider three cases: stationary (no mobility), 30 and 50 km/h.

6 Wireless-to-wireless VoIP measurement results

Our study is conducted in Taipei City, where the RF environment is affected by tall buildings and heavy vehicle traffics, and more serious interference is observed as compared with the line-of-sight environment. Every stationary test is conducted for 5 min, and every mobility test is conducted for 2 min. During a stationary test, roughly 2 400 000 bytes were sent in one-way VoIP link. Similarly, in a mobility test, we measured roughly 960 000 bytes for one-way VoIP link. Therefore the equivalent bandwidth consumed is about 80 kbps, which is higher than the default G.711 data rate (i.e. 64 kbps) because of extra RTP header overhead.

This section shows the effects of CPE mobility and background traffic on MOS, packet loss, packet delay and

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jitter. Figs. 7–10*a* show the expected MOS values. Figs. 7–10*b* and 10*d* give the real-time measures of an example experiment.

6.1 Mean opinion score

Fig. 7 shows the MOS performance. Fig. 7*a* indicates that the average MOS values for stationary CPEs are above 4.0, and are larger than 3.9 for moving CPEs. The MOS slightly decreases as the CPE speed increases. The MOS values are insignificantly affected by the background traffic.

Figs. 7*b*–*d* illustrate real-time MOS measurements of CPE1 (the grey curve) and CPE2 (the black curve) in a typical experiment, where the uplink background traffic of CPE3 is 3 Mbps. When both CPEs are stationary, the real-time MOS values are measured for 5 min. In this case, the MOS is typically maintained higher than 3.8, and the values of most MOS drops are still above 3.0. These MOS drops are partly because of tall buildings and the street traffic surrounding the minivan of the CPEs. The 5-min average MOS values are 4.32 at CPE1 and 4.3 at CPE2.

In every mobility test, the real-time MOS values are measured in 2 min. At the speed of 30 km/h, a handover occurs roughly at the 80th second. The real-time MOS may drop significantly from 4.35 to 1.0 (for CPE1) and 2.1 (for CPE2) as illustrated in Fig. 7*c*. The average MOS values are 3.83 (for CPE1) and 3.96 (for CPE2). At the speed of 50 km/h (Fig. 7*d*), after the first handover occurring at the 50th second, the MOS values become very unstable, and the MOS is not recovered back to 4.34. The 2-min average MOS values are 3.97 (for CPE1) and 3.79 (for CPE2), respectively.

Our study indicates that the CPE mobility does not degrade the MOS performance except when the handovers occur.

6.2 Packet loss

Fig. 8*a* illustrates the average packet loss. For stationary CPEs, the average packet loss is less than 0.01%. For moving CPEs, the packet loss is less than 0.7%. The packet loss increases as the CPE speed increases.

The packet loss of stationary CPE is not affected by the background traffic. On the other hand, the background traffic significantly affects the moving CPEs. At CPE speed of 30 km/h (50 km/h), the packet loss increases from 0.325% (0.375%) to 0.566% (0.675%) when the background traffic increases from 0 to 3 Mbps.

Like Figs. 7*b*–*d*, Figs. 8*b*–*d* illustrate an example of real-time packet loss measurements with various CPE speeds. When both CPEs are stationary, most packet loss values are less than 0.064%. The 5-min average packet loss value is 0.013%.

At the speed of 30 km/h (Fig. 8*c*), the 2-min average packet loss values are 0.783% (for CPE1) and 0.35% (for CPE2). At the speed of 50 km/h (Fig. 8*d*), the 2-min average packet loss is 0.7% (for CPE1) and 0.65% (for CPE2), respectively.

We note that packet loss in bursts is more damaging than uniform random packet loss. The voice quality is affected when consecutive five or more packets are lost at a time. Fig. 8*b* shows that for stationary CPEs, the maximum number of consecutive lost packets is 1. For CPEs moving at 30 km/h (Fig. 8*c*), packet loss in bursts are observed when handovers occur, and the maximum number of consecutive lost packets is 3 occurring at handover. For CPEs moving at 50 km/h (Fig. 8*d*), lost packets in bursts are more serious. The maximum number of consecutive lost packets is 3. In our experiments, the packet loss measures satisfy the requirement of WiMAX forum (i.e. less than 1%).

6.3 One-way packet delay

In Fig. 9*a*, the average one-way packet delay (including 10 IP hops and two WiMAX radio links) is less than 45 ms for stationary CPEs, and is less than 52 ms for moving CPEs. The delay increases as the CPE speed increases (because of the handover impact).

When the background traffic increases, the one-way packet delay tends to increase for moving CPEs. The background traffic effect on stationary CPEs is negligible.

Figs. 9*b*–*d* illustrate an example of real-time packet delay measurements with different CPE speeds. When both CPEs are stationary, packet delays are always less than 63 ms. The 5-min average packet delay is 44 ms at CPE1 and 40 ms at CPE2.

At the speed of 30 km/h (Fig. 9*c*), most packet delays are less than 88 ms, and the maximum packet delay is 132 ms occurring at the handover. The 2-min average packet delay is 51 ms (for CPE1) and 50 ms (for CPE2). At the speed of 50 km/h (Fig. 9*d*), most packet delays are less than 100 ms, and the maximum packet delay is 289 ms. The 2-min average packet delay is 55 ms (for CPE1) and 49 ms (for CPE2).

In our experiments, most packet delays are much less than the acceptable upper limit of packet delay (i.e. 150 ms).

6.4 Jitters

Fig. 10*a* shows that the average jitter is less than 4.3 ms for stationary CPEs, and is less than 6 ms for moving CPEs. Our experiments indicate that for stationary CPEs, the background traffic does not seem to affect jitter. For moving CPEs, jitter increases as the background traffic increases. However, it is not clear why CPE speed at 30 km/h tends to have the worst jitter performance (such phenomenon was also observed in other experiments).

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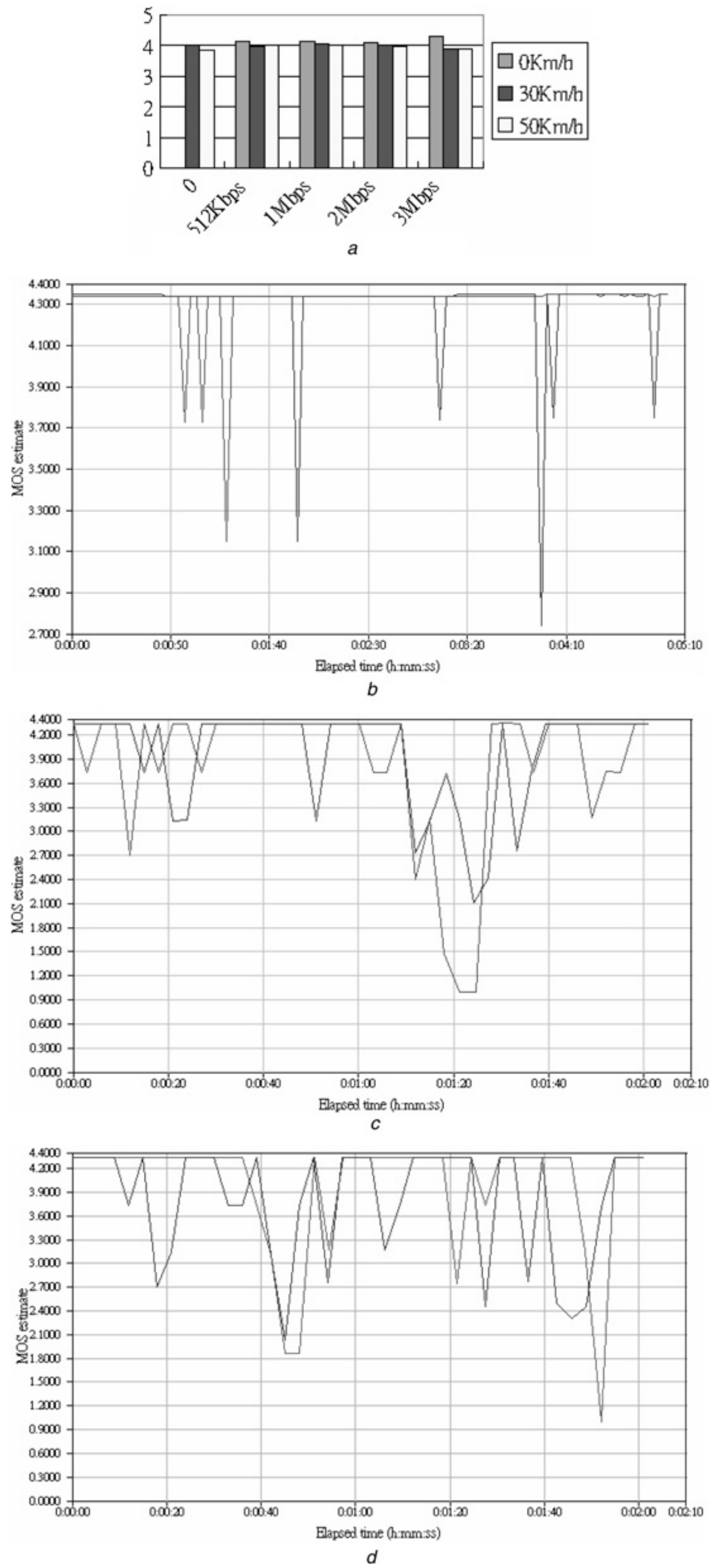

Figure 7 *MOS measurements*

a Average MOS
b CPE speed: 0 km/h (uplink background traffic: 3 Mbps)
c CPE speed: 30 km/h (uplink background traffic: 3 Mbps)
d CPE speed: 50 km/h (uplink background traffic: 3 Mbps)

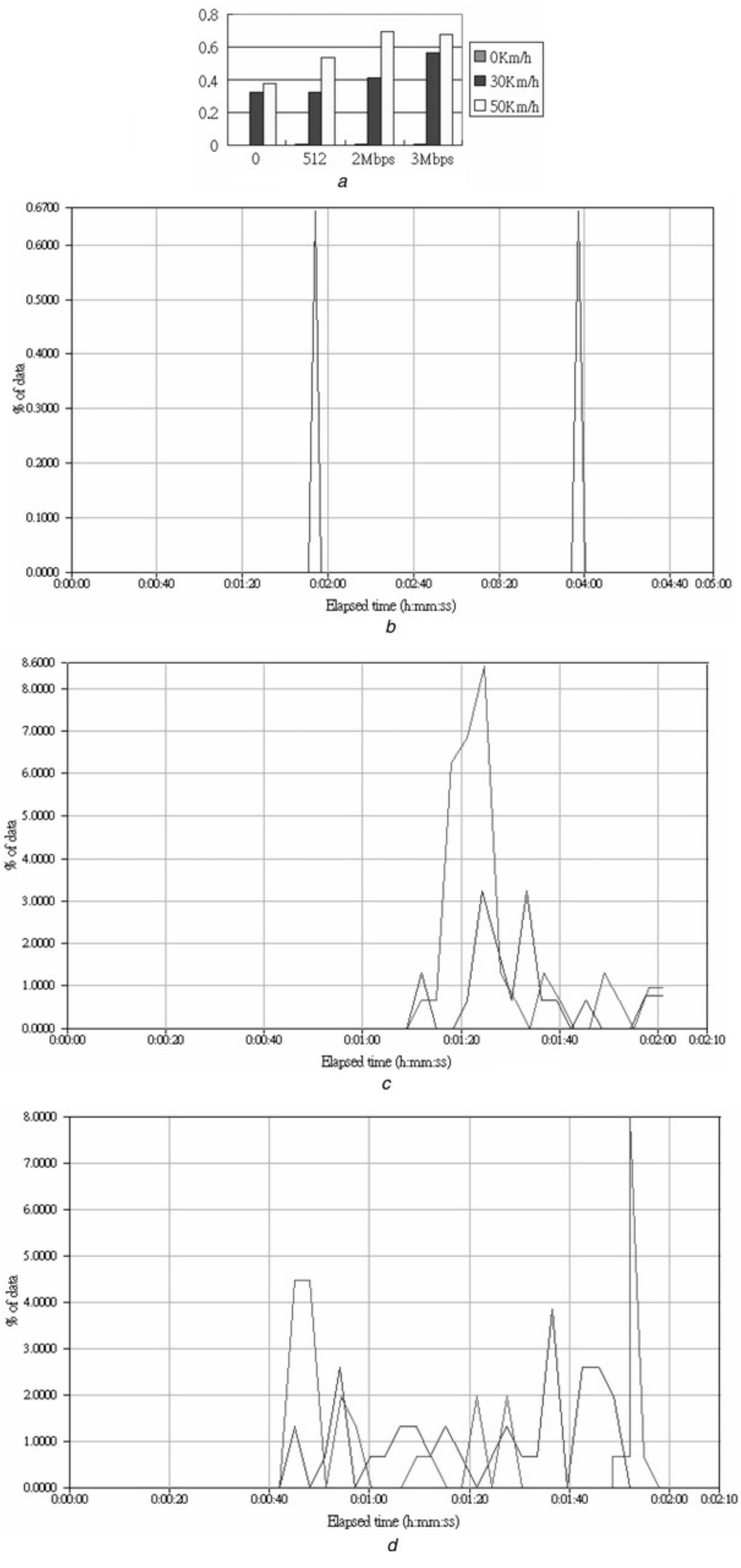

Figure 8 *Packet loss measurements*

a Average packet loss (%)
b CPE speed: 0 km/h (uplink background traffic: 3 Mbps)
c CPE speed: 30'km/h (uplink background traffic: 3 Mbps)
d CPE speed: 50 km/h (uplink background traffic: 3 Mbps)

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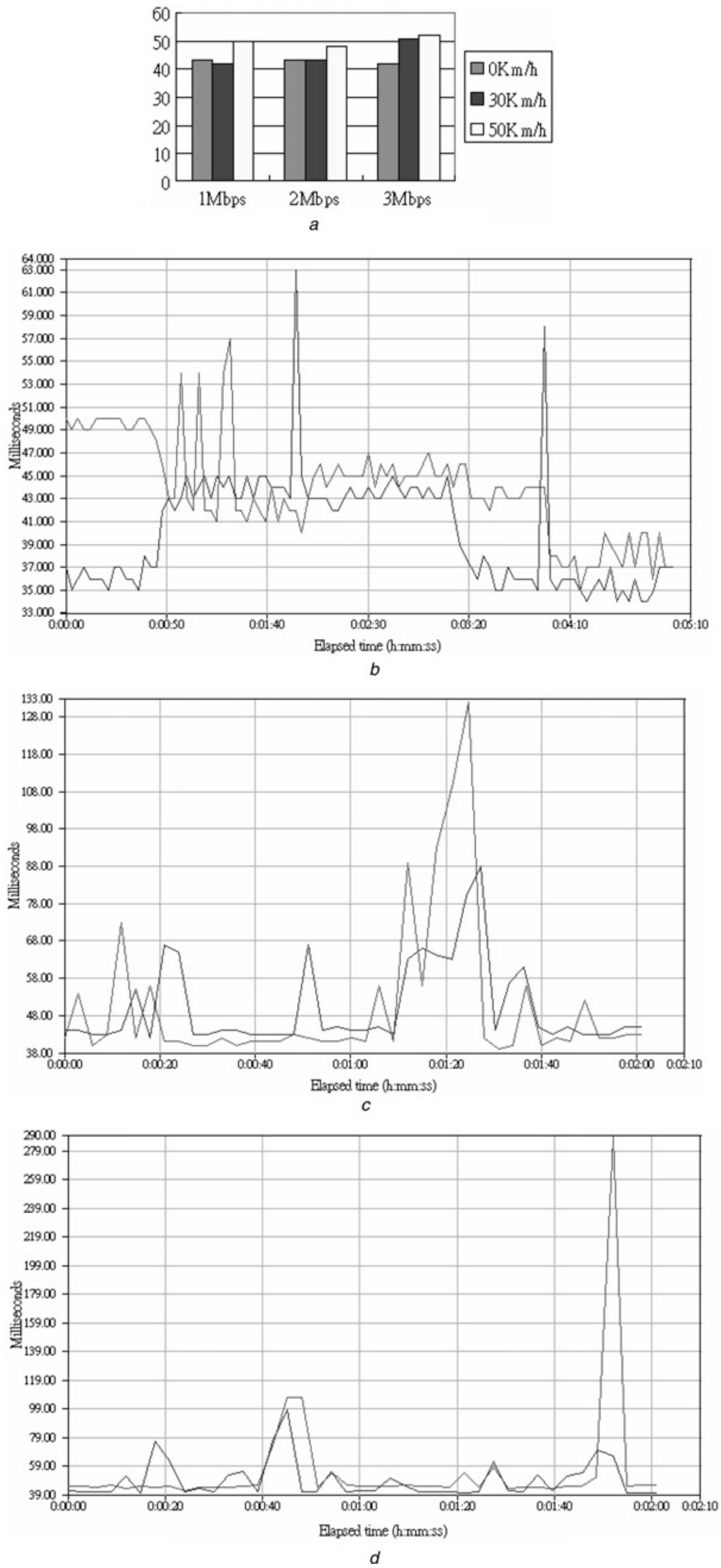

Figure 9 *One-way packet delay measurements*

a Average one-way packet delay (ms)
b CPE speed: 0 km/h (uplink background traffic: 3 Mbps)
c CPE speed: 30 km/h (uplink background traffic: 3 Mbps)
d CPE speed: 50 km/h (uplink background traffic: 3 Mbps)

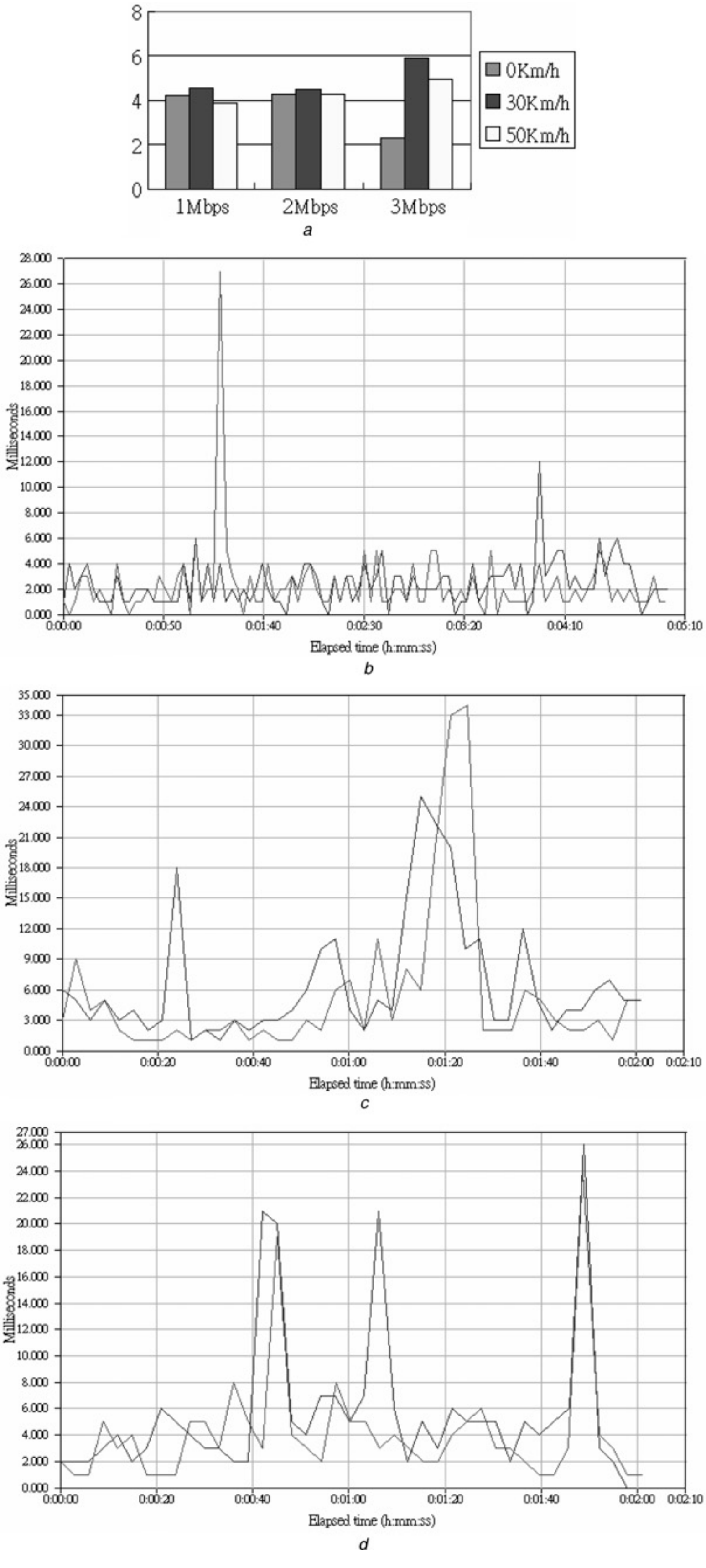

Figure 10 *Jitter measurements*

a Average jitter (ms)
b CPE speed: 0 km/h (uplink background traffic: 3 Mbps)
c CPE speed: 30 km/h (uplink background traffic: 3 Mbps)
d CPE speed: 50 km/h (uplink background traffic: 3 Mbps)

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Figs. 10*b*–*d* give examples of real-time jitters measurements. When both CPEs are stationary, all jitter values are less than 27 ms. The 5-min average jitter is 2.23 ms at CPE1 and 2.42 ms at CPE2.

At the speed of 30 km/h (Fig. 10*c*), the maximum jitter is 34 ms when the handover occurs. The 2-min average jitter values are 5.225 ms (for CPE1) and 6.6 ms (for CPE2). At the speed of 50 km/h (Fig. 10*d*), maximum jitter is 27 ms. The average jitter values are 4.2 ms (for CPE1) and 5.75 ms (for CPE2). After the handover, jitters occur in bursts. Figs. 10*c* and *d* show that jitter is more seriously affected by handover at 30 km/h than that at 50 km/h.

The study also indicates the following observations:

- The impact of background traffic on VoIP is mostly insignificant.

- The MOS values slightly decrease as the CPE speed increases. The MOS values are not affected by the background traffic.

- The packet loss increases as the CPE speed increases. The packet loss of stationary CPE is insignificant, and is not affected by the background traffic. On the other hand, the background traffic significantly affects the moving CPEs.

- The one-way packet delay increases as the CPE speed increases. The background traffic slightly affects the packet delays for moving CPEs. The background traffic effect on stationary CPEs is negligible.

- Impacts of CPE speed and background traffic on the jitters are not clear in our study. However, all experiments indicate resilience against jitters.

- The values of all jitter samples observed in our study are much lower than the unacceptable jitter value (i.e. 25 ms).

7 Conclusions

Our investigation upon the experimental results indicates the performance of a VoIP service using the WiMAX-based infrastructure of the M-Taiwan Program conforms very well to the standard requirements of G.107 under the worse-condition and stringent scenario where both VoIP CPEs are wirelessly connected to the same WiMAX BS with both moving CPEs at the speeds up to 50 km/h while both going under handovers at the same time.

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