Experimental Analysis of VoIP over Wireless Local Area Networks

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***Abstract*—VoIP is a rapidly growing technology that enables the transport of voice over data networks such as the public Internet. VoIP became a viable alternative to the public switched telephone networks (PSTNs). In parallel, a dramatic increase is happening in the deployment of Wireless Local Areas Networks (WLAN) in buildings and corporate campuses. Nowadays, WLAN is mostly used for ordinary data services such as web browsing, file transfer and electronic mail. However, with the emerging usage of VoIP telephony, WLAN are sought to be used as an access infrastructure for enabling such applications. One of the issues of using VoIP over WLAN is the effects caused by users roaming within and between WLAN subnets during a VoIP session. The latency and the jitter are greatly impacted when the control of the mobile node is handed over from one access point (AP) to another one. This poses a challenge to providing and preserving QoS for VoIP users in WLAN environments. In this paper, we propose to study and measure the effect of the handover for both intra and inter mobility for VoIP traffic.**

***Index Terms*—VoIP, WLAN, Handover, Voice quality, Wireless mobility**

I. INTRODUCTION

Voice of IP (VoIP) is a rapidly growing technology that enables the transport of voice over data networks such as the public Internet. VoIP became a viable alternative to the public switched telephone networks (PSTNs), and it is increasingly deployed on corporate environment and campuses. It uses a number of protocols which ensure that voice communication is appropriately established between parties, and that voice is transmitted with a quality close to that we are accustomed to in the PSTN. VoIP uses signaling protocols such as the Session Initiation Protocol (SIP) [1] and H.323 [2]. Concurrently, in the access technology used for IP-based networks, a rapid and wide deployment of wireless local area networks (WLAN) is taking place in most corporate buildings, small offices and home offices (SOHO) as well as public spaces such as commercial malls and airports. WLAN technology is based on the IEEE802.11 network access standards [3]. The use of WLAN enables users to have instant access to the Internet services regardless of their location in the network. In addition, connectivity is continuously offered to the users while roaming from one place to another [4]. As the user moves from one radio coverage to another, the mobile device transfers its control between the Access Points (AP). This transfer process is called handover or handoff. The performance of certain applications can be impacted during a handover.

VoIP is a service that has stringent QoS requirements as to the timeliness and the quality of the voice required for users in WLAN-based access networks [5, 6]. Several studies have shown that mobility handover can have an impact on the quality of the voice due to the delays caused by the various operations executed during the handover [7, 8 and 9].

In this paper, we propose to study the effect of the mobility handover on VoIP communications. Our work focuses on the impact of handoff mechanism on the objective quality of the voice traffic, the transmission delays, and delay jitter. The remainder of the paper is organized as follows. In section II we discuss the QoS requirements of VoIP. In section III, we discuss the problem of handover and its impact on VoIP QoS in the context of WLAN. To validate our analysis, we present some experiments and discuss the respective results in section IV. Finally, we conclude this paper in the last section.

II. QOS REQUIREMENTS

It is crucial to the success of deploying VoIP applications over WLAN to have the ability to support and provision QoS capabilities [9]. Furthermore, voice services inherently involve call control signaling that requires a high level of priority in order to meet the

Based on “Study of the Effect of Mobility Handover on VoIP over WLAN”, by A. Lakas, M. Boulmalf which appeared in the Proceedings of the IEEE International Conference on Innovations in IT, 2006, Dubai, UAE, November 2006. © 2006 IEEE.

timing constraints of interfaces to external networks, such as the wireless cellular network or the public switched telephone network (PSTN).

TABLE II.
CODEC REQUIREMENTS

Coding	Algorithm	Band-width (Kbps)	Sample (ms)	Typical IP bandwidth (Kbps)
G.711	PCM	64	0.125	80
G.723.1	ACELP	5.6	30	16.27
G.723.1	ACELP	6.4	30	17.07
G.726	ADPCM	32	0.125	48
G.728	LD-CELP	16	0.625	32
G.729	CS-ACELP	8	10	24

For network planners who are deploying a VoIP over WLAN application, one of the first issues to be addressed should be network capacity. To ensure the network is able to deliver the required QoS capabilities for a voice application, designers must anticipate and analyze how the WLAN will be used. Several questions, such as the following, must be answered:

- What types of QoS capabilities will be deployed?
- How much network capacity must be set aside for these QoS capabilities?
- What is the projected growth rate for QoS capabilities on the WLAN?

The above-mentioned questions are network-design-specific considerations for VoIP and other services requiring QoS capabilities.

In wireless networks, voice is digitized with the G.711 coding standard and transported at 64 Kbps. While G.711 is the main digital codec for toll quality voice services, a number of more efficient codecs are used for both cellular and voice applications. In IP networks, voice codecs are placed into packets with durations of 5, 10 or 20 msec of sampled voice, and these samples are encapsulated in VoIP packets. Table II above illustrates the various codecs [10] and their corresponding bandwidth requirements for IPv4.

A. Delay and Jitter Requirements

In addition to the overhead incurred by the voice compression, and regardless of the application, the time delay and jitter of the VoIP will be a design consideration [11]. The two following issues relate to time delay and jitter: (1) Signaling for call set up, tear down and other call control communications will be delayed. Worst case delay is the principal concern; (2) the jitter in the voice traffic/bearer channel will cause delay.

VoIP signaling and voice traffic are not separate communications channels. VoIP packets exist as virtual communications within a single channel. Only queuing priorities can ensure timely delivery of voice packets when other types of packets are competing for services in an IP network. This situation is complicated when a wireless user is moving and there is access point-to-access point handover in the network. Further delays are added into the WLAN as the user must associate with an access point, authentication must take place and the handover must be completed. Table I lists the voice delay requirements as specified by the G.113 [12]:

TABLE I.
G.113 DELAY SPECIFICATION

Delay	Quality
0 to 150 *msec*	Acceptable to most applications
150 to 400 *msec*	Acceptable for international connections
> 400 *msec*	Acceptable for public network operation

B. QoS and Handover

A handover occurs when the mobile device's physical connection changes from one access point to another within the same domain. This operation consists of the following:

1) Releasing the connection from the old access point
2) Establishing the connection to the new access point
3) Updating the binding between the mobile device's IP address and its temporary Layer 2 ID, on the subnet access router.

The above steps are not necessarily executed in the order shown. If step (1) is executed before (2), this is referred to as "break-before-make" handover. If step (2) is executed before step (1), this referred to as "make-before-break" handover. The break-before-make handover may result in connectivity disruption and/or loss of data. Therefore, transmitted voice packets may be either delayed or discontinued for the period of the handover resulting in an increase of the transmission jitter. Because the two access points belong to the same coverage domain (i.e. the same subnet), the handover allows the mobile device to preserve the same IP address. It has no involvement from outside the subnet and therefore is usually fast enough to not incur noticeable delays. Figure 1 illustrates the various operations executed during an intra-domain handover.

When the mobile device is moving away from its current access point, the signal quality received by the access point will decrease down to a certain signal quality threshold (Table III). At this point, the mobile device triggers the handover procedure. This procedure consists of looking for a better access point to re-associate with so that the mobile device is better served.

The signal quality is typically measured in term of the signal-to-noise ratio (SNR). Therefore the decision of initiating a handover is based on two parameters specified in the IEEE802.11 standards: the cell search threshold and the delta SNR value measured in decibels (dB) -- see table below. When the SNR value reaches the cell search threshold, the device starts the search phase explained earlier.

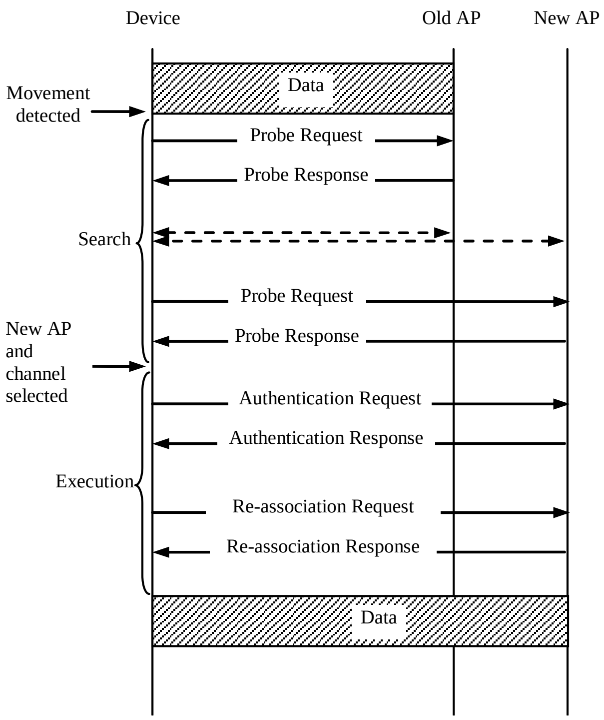

Figure 1. Handover procedure.

The search phase consists of the following:

1) Interrupting any data transfer, whether downstream or upstream
2) Scanning all the existing access points and in all the channels looking for a channel on an access point that can serve better the device
3) The selection of the channel on the new access points is subjected to criteria that are explained further next.
4) When the mobile device finds an access point with a signal value above the threshold, and if the difference between the SNR of the old access point and that of the possible new one is above the delta SNR value, then it initiates the actual handover. This procedure is illustrated in Figure 2.

TABLE III.
IEEE802.11 HANDOVER THRESHOLDS

Threshold	Access point density		
	Low	Medium	High
Cell Search (dB)	10	23	30
Delta SNR (dB)	6	7	8

The actual handover itself is finalized with a re-association phase where the mobile device is now controlled by the newly selected access point; i.e., all incoming and outgoing traffic is transmitted via the new access point. However, the handover is not completed successfully until other steps are carried out. Depending on the configuration of the access point, the handover may require further steps such as authentication and QoS profile exchange. During the authentication phase, information about the device and the access point credentials is exchanged in order to allow the connection. Other procedures may be included during the handover. IEEE 802.11 working groups are still finalizing the standard about QoS (802.11e), access point interoperability (802.11f) and security (802.11i).

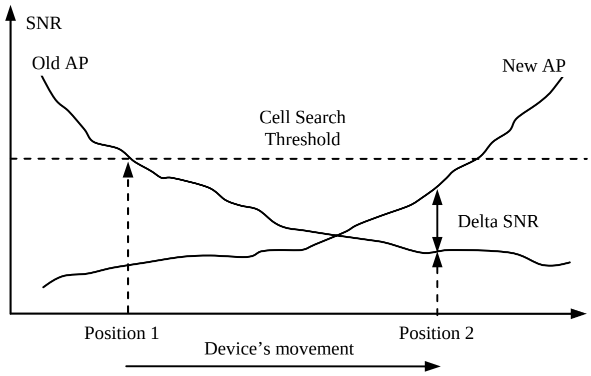

Figure 2. Handover decision parameters.

III. VoIP OVER WLAN AND QoS

The IEEE 802.11 standard specifies that a mobile device can only be associated with one AP at a time [3], so there is a risk that the communication is interrupted while performing the handover. The duration of the period when the mobile device is unable to exchange data traffic via its old and new access points is often referred to as the handover latency or handover delay. If the mobile device experiences degraded signal quality in the communication with its access point, it will at some point in time trigger a handover procedure. If the handover threshold value is configured so that a handover is triggered before connectivity with the current access point is lost, then the time to detect movement will not affect the total handover latency. To find candidate access points to re-associate with the mobile device will start to scan the different radio channels.

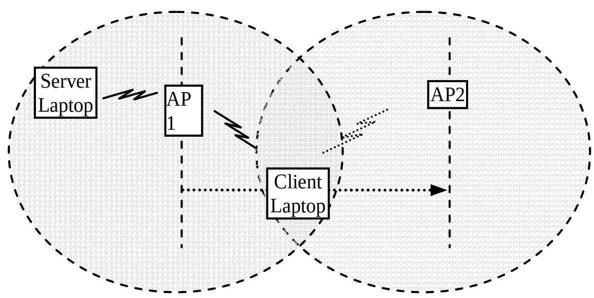

Figure 3. Experiment setup.

Since 802.11 networks were designed to carry data, not voice, 802.11 b/g standards have no QoS mechanisms built-in to tell the network to prioritize voice packets over data, so a surge in network traffic can disrupt voice calls. Given the fact that voice is a real time application, QoS control is essential and without it may lead to end-to-end delays, jitter, out of sequence errors, packet losses and contention (resulting in people talking over each other or the sound breaking up).

IV. Experiments

A. Experiment Setup

The setup of the experiment consists of closed network containing two Linksys access points supporting 802.11g and two Windows based laptops, each equipped with 802.11g LAN card. The setup is illustrated in the diagram depicted in Figure 3. We have used LanTraffic as a traffic generator and analyzer which has two program components: one running on the server laptop, and one running on the client laptop. The server program generates traffic destined to the client laptop. In these experiments, we have focused on the observation of the SNR, the traffic throughput and the delay jitter. We also analyzed the correlation between these parameters. In addition, we have used WinSIP and WinEyeQ (from Touchstone Inc.) in certain experiments.

We have run three experiments:

- *Experiment_1*: in this scenario, we have considered the effect of the inter-domain handover on the variation of the SNR during the handover procedure.
- *Experiment_2*: in this scenario, we have considered the effect of the handover on the traffic throughput.
- *Experiment_3*: in this scenario, we measured the effect of the handover procedure on the delay jitter.

- *Experiment_4*: in this scenario, we tested the effect of the number of active calls on the delay jitter.

- *Experiment_5*: in this scenario, we considered the use of various voices codecs and their effect on the MOS. The codecs used are G.711 (64kpbs), G.729 (16kbps), G.728 (8kbps) and G.723 (5.8kbps).

- *Experiment_6*: in this scenario, we have tested the effect of the number of active calls on the MOS using G.711 and G.723 codecs.

- *Experiment_7*: in this scenario, we have tested the

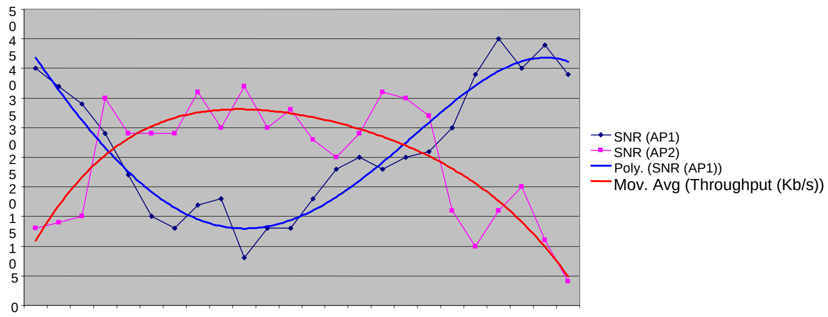

Figure 4. SNR variation during handover

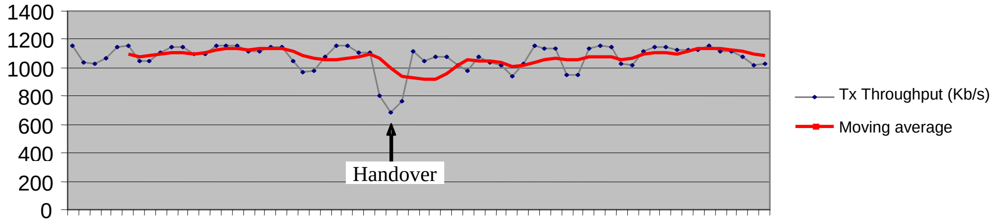

Figure 5. Effect of the handover on the traffic throughput

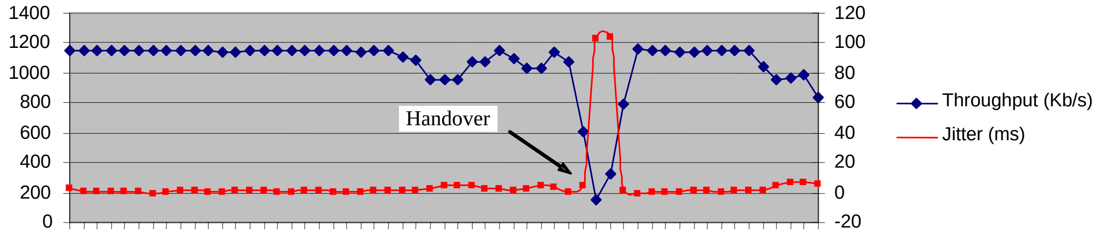

Figure 6. Effect of the handover on the jitter

effect of the number of active calls on the packet loss G.723 codecs.

B. Result Assessment

Following are the results of the experiments above:

1) Experiment_1: the result of this experiment, illustrated in the chart below, shows the SNR variations of access point AP1 and AP2. SNR(AP1) decreases as the client laptop moves away from AP1. Concurrently, SNR(AP2) increases as the client laptop gets closer to AP2. The chart also includes the trend of the SNR variation in Red and Blue respectively, for AP1 and AP2.

2) Experiment_2: the result of this experiment, illustrated in Figure 5, shows the traffic throughput variation. The chart indicates that the throughput experiences a brief dip during the handover. This sudden drop amounts to a difference of 450 Kbps. This can be explained by the decrease of the bandwidth caused by the packet loss experienced during the handover procedure.

TABLE VI.
VOICE CODECS VS. MEAN OPINION SCORE

Audio Codec Format	MOS
G711 (64kbps)	4.195
G728 (16kbps)	4.035
G729 (8kbps)	3.945
G723 (5.3kbps)	3.613

3) Experiment_3: In this experiment, we focused on the analysis of the jitter variation and its relationship with the variation of the throughput. The result shown in Figure 6 indicates a brief, but big spike, in the delay jitter when the handover occurs. The spike amounts to 100 ms during a short period of 5 ms. The jitter spike is accompanied with a considerable drop in the throughput which amounts to 1Mb/s. Also, the throughput averages before and after the handover period is comparable to that of the previous experiment.

4) Experiment_4: In this experiment, we varied the number of active voice sessions and examined the effect on the jitter. In order to analyze also the effect of the codec used, we tried two codecs" G711 (5.3kbps) and G723 (64kbps). The result shown in Figure 7 indicates, for both types of codecs, we start observing an increase in the jitter beyond a number of sessions of 20 calls. However, the jitter increase for G723 is a lot faster than that for G711. This is easily explained by the bandwidth required by each codec. At 50 active calls, the hitter is measured to be 160ms for G723 and 62ms for G711.

5) Experiment_5: In this experiment, we assessed the effect of the use of various voice codecs on the subjective quality of the voice. As indicated by Figure 8, the quality decreases as we go from G.711 till G.723. This is explained by the bandwidth required by each codec. For instance G.711 requires 64kbps whereas G.723 requires only 5.8kbps and therefore the loss of the information in a single frame.

In addition, we suspect that with a higher number of active calls competing on the existing bandwidth, as in this experiment, more information is lost due to packet loss as seen in experiment_6 and experiment_7.

TABLE V.
ACTIVE CALLS VS. MEAN OPINION SCORE

Number of calls	MOS (G711)	MOS (G723)
10 calls	4.195	3.613
20 calls	4.195	3.5605
30 calls	3.613	3.1859
50 calls	3.613	2.31232

6) Experiment_6: This experiment contributes to clarify certain results in the previous experiments. Here, the results indicate that as the number of active numbers increases the quality of voice decreases. However, two elements need to be explained; first, that both G711 and G.723 maintain the same MOS until the number of active calls reaches 20 calls. The decrease in the MOS starts from then on. Keeping invariably the same MOS up to 20 calls is explained by the absence of any other background traffic. Calls are competing only between themselves. Second, beyond 20 calls, the decrease in MOS is more dramatic for G.723 than it is for G.711 for the same reasons referred to in the previous experiment.

TABLE IV.
ACTIVE CALLS VS. PACKET LOSS

Number of calls	Packet Lost
10 calls	0.0%
20 calls	1.0%
30 calls	4.9%
50 calls	10.7%

7) Experiment_7: Packet loss is a major factor in degrading the quality of voice. The results depicted in Figure 10 indicate clearly an increase in the number of packet loss as the number of active calls is increased. A good correlation between this experiment and the previous ones is shown in these results. One may notice that the increase in the packet loss up to 20 calls is relatively small. However, beyond 20 calls, the increase is very apparent. Despite the minimum framing rate used by the voice codec G.723 in this experiment, the degradation in the voice quality

incurred by the higher number of calls competing for the medium is straightforward.

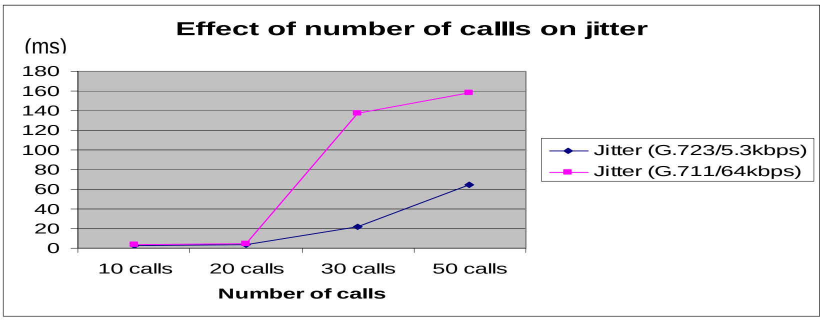

Figure. 7. Effect of the number of active calls on the jitter

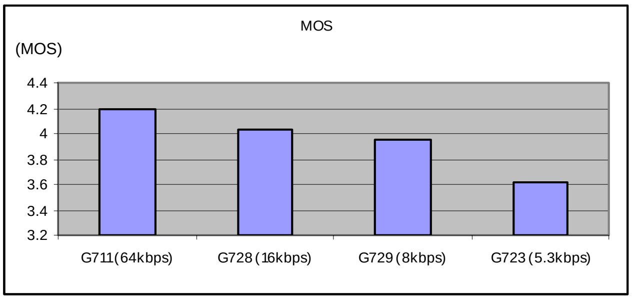

Figure. 8. Effect of codec choice on the mean opinion score

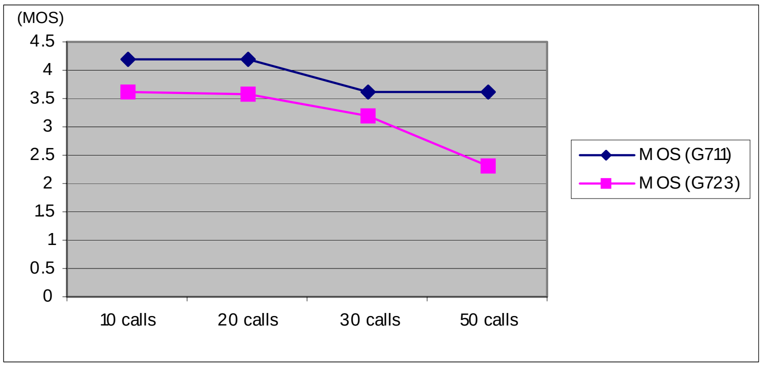

Figure. 9. Effect of the number of active calls on the mean opinion score

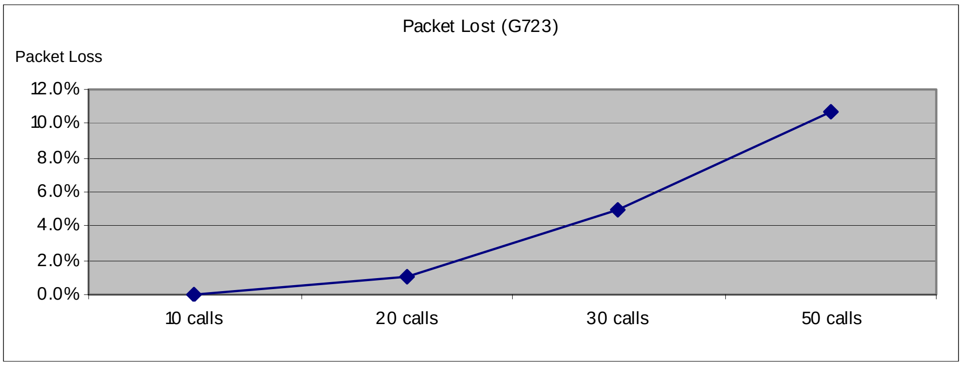

Figure. 10. Effect of the number of active calls on the voice packet loss

V. Conclusion

In this report, we have presented a study on the impact of the handover on the quality of VoIP over WLAN. The study was oriented towards the assessment of the variation of the throughput and the packet delay jitter during the handover operation. The results presented in this show the effect of the handover on the voice transmission over an 802.11 based LANs. Despite the fact that the handover configuration in our experiments does not include extra operations related to the authentication, encryption information exchange and QoS parameters transfer, the results indicate that intra-domain handover can still impact the quality of voice through the jitter increase and the drop in the throughput. The latency incurred in re-establishing the forwarding path between the mobile device and the new access point decreases the VoIP quality. Therefore, new methods for intra and inter-domain handover are required. These methods should keep the latency to an acceptable minimum before VoIP can be successfully deployed at a large scale. Ongoing standard activities are aiming towards filling this gap, but acceptable solutions are yet to be found for responding to VoIP requirements.

Acknowledgment

The authors wish to thank Mr. Saad Elias, Dr. Adel Serhani for their help and assistance in realizing this project. We are grateful to the staff of Touchstone Inc. for their assistance and support. This work was financially supported by the Research Affairs of the UAE University under a contract no. 05-03-9-11/05.

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