Impact of Wireless Devices over Real-time Applications: An Empirical Test-bed Analysis

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***Abstract*—Wireless Local Area Network (WLAN) has enabled greater communication capabilities compared to LAN counterpart. However, when it comes to quality of service (QoS), WLAN has lower reliability, where it has higher latency and packet-loss, especially for real-time streaming applications such as voice over internet protocol (VoIP) and multimedia. Thus, improving QoS in WLAN is a major research challenge, where packet-loss is considered most critical while other factors, such as latency and jitter are acceptable to certain level of tolerance. Theoretically WLAN can provide higher bandwidth and lower latency; however, practical results disagree with this statement. In this paper, we investigated the dilemma using different combination of wireless devices within a local area network to identify the problem. We propose an infrastructure based scheme to improve the QoS for real-time traffic over WLAN. We setup a wireless network using different types of wireless devices and tested VoIP traffic over it. Our simulation results for G.729.2 codec on IPv4 and IPv6 using virtual access point (v-AP) showed better performance than physical access point (ph-AP). We also observed that usage of different wireless devices within a network highly reduced real-time application performance.**

Keywords: Delay, Jitter, VoIP traffic, Packet-loss, RTT, Wifi, virtual AP, and wireless 802.11g.

I. INTRODUCTION

Voice over internet protocol (VoIP) is a real-time application which allows users to establish a voice communication over packet-switched networks (Internet). In the past few years, VoIP has gained huge popularity around the world due to the reduced cost associated with using VoIP compared to public switched telephone network (PSTN) [1-3]. Nowadays usage of VoIP over WLAN in home and office environments is becoming more common as new mobile devices have WLAN connectivity. These devices have enabled users to be connected to access points (APs) via WLAN for better signal strength and have provided multiple users to have different types of communication options such as data, voice, video, sensor etc. However, when it comes to QoS, in particular for real-time applications such as voice and multimedia, there are concerns in regards to packet-loss and latency [9]. QoS is a vital factor where a good quality, e.g., for VoIP can be measured if round trip time (RTT) delay is lesser than 150 milliseconds (ms) and packet-loss is < 3% [4, 10].

According to study by Zheng et al., [4], in 2009, Skype had more than 80 million subscribers and such services were becoming more frequently used as it was accessible from modern wireless-enabled devices such as smart phones and tablets which have built-in VoIP features. In line with the widespread use of VoIP, new methods and algorithms have been developed to enhance the QoS for VoIP communications [2-4, 16]. However, these new devices are mostly based on wireless which reduces the QoS of VoIP over WLAN [5-6]. In the case scenario presented by Shin and Schulzrinne [7], it was observed that LAN capability of handling QoS with heavy background load was much stronger and reliable than WLAN; especially when it comes to comparing the performance of real-time applications.

Internet users around the globe are rapidly increasing and more mobile devices are being connected to the WLAN; another great challenge ahead for internet services is the transition from IPv4 to IPv6. The larger number of desktops, laptops, mobile devices and other computing machines requiring IP addresses for access to internet and networking has been well established [27]. However, growth of IPv4 rapidly increasing and soon, all available IP addresses will be exhausted. To resolve this issue, Internet engineering task force (IETF) has designed a new version of IP protocol known as IPv6 to fulfill the current and future needs of IP addresses. IPv6 was designed with enhanced features and main advantage of IPv6 is its ability to support large numbers of addresses (2^128-bit). Many IPv4 based solutions, methods and algorithms were proposed over a decade ago to enhance the quality of internet services; however, new issues are expected to arise when these legacy services are moved to perform on IPv6 [27].

In this study, we proposed an infrastructural based solution to improve the QoS for real-time applications over WLAN. Our scheme was designed for real-time applications; however, only VoIP traffic has been tested yet. To test VoIP, we have selected most commonly used voice codec known as G.711.1, G.723.1 and G.729.2 while for WLAN we selected IEEE 802.11g protocol. We also applied our designed scheme on both the IPv4 and IPv6 protocols to identify the differences between their performances within our and thus to aid the impending switch from IPv4 to IPv6 networks.

The organization of this paper is as follows. The next section covers background and related works related to our study. Section 3 and 4 presents the procedure of network test-bed and information regarding network simulation tool, respectively. Section 5 details the network designed environment and Section 6 presents the experimental results. Finally, section 7 outlines the discussions followed by conclusions in Section 8.

II. Background and Related Works

VoIP, also known as IP telephony, is a voice communication system which allows a user to have voice conversation via broadband connection and a phone set that converts analog voice to digital packets. There are various voice protocols and codecs which carry digital packets via the Internet and deliver them to their destination. Figure 1 illustrates the process of voice being transferred over the internet [11].

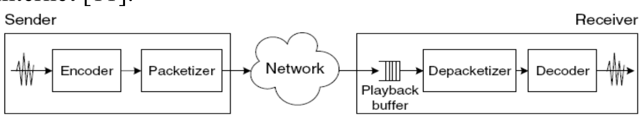

Figure 1: End-to-end components of VoIP [11]

VoIP is a real-time application which is highly delay sensitive compared to data traffic. QoS for VoIP is much better on LAN than on WLAN due where wireless technologies have their own reliability factors. According to [12] there is certain level of latency which is tolerable for VoIP. Table 1 below illustrates the 3 stages of quality assurance for VoIP's capacity.

Table 1: QoS Requirements for VoIP [12]

Quality	**Delay (ms)**	**Jitter (ms)**	**Loss**
Good	0 – 150	0 – 75	< 3%
Medium	150 – 400	0 – 125	< 7%
Poor	<400	0 – 225	>7%

There is a collection of voice codecs such as the G.711 and G.7xx series developed by international telecommunications union (ITU) for audio compression and de-compression. Each codec has different packet sizes and performs differently. In this research we have selected most commonly used voice codec known as G.711 [13]. The G.711 codec has high bit rate (64Kbps) of the ITU standards and it is used for digital telephony networks and IP networks. Theoretically G.711 codec provides good quality of voice and requires low processor but it requires greater bandwith [14].

Virtual Access Point (v-AP) is a latest feature which supports most of network cards. This feature allows users to turn their WiFi network cards into virtual APs. Hence, multiple connections can be established via V-AP. Most recently developed/released operating systems have this feature built-in; however, it requires set of commands to be executed in order to enable this mode. There are few software available in the market that supports graphical user interface (GUI) mode and also provide a user guide to operate v-AP different setting [15].

WLAN is based on IEEE 802.11 standards and has many editions such as IEEE 802.11a, b, g, and n; each edition works on different frequency and has valuing bandwidth limitations.

Recent studies have investigated the use of dual-WLAN for improving QoS [20-22]. In [20] the authors have mentioned how important it is to have high reliability and low delay in wireless communication. They proposed a dual-channel wireless model which broadcasts same packets using multiple transmitters. Their simulation results indicated that if bit error rate (BER) is set to (4×10^{-4}), it can provide less delay using dual-channel. Also, they showed that dual-channel reduces packet-loss when compared to a single channel.

In an earlier study [23] VoIP and UDP traffic was tested over IEEE 802.11b. Main focus of the experiment was to identify the maximum simultaneous VoIP calls over IEEE 802.11b with excellent QoS. Thus, ITU G.711a-Law codec was transmitted with the speed of 10ms data rate and parameters such as packet-loss, delay and jitter were considered. The outcome clearly showed that five number of concurrent voice calls were established while having 0% packet-loss and ~ 5ms RTT and ~7ms jitter. However, when a sixth simultaneous call was established slight increase in RTT was found while seventh simultaneous call caused some amount of packet-loss and marginally added more RTT. Hence, concluded that maximum number of concurrent voice calls over IEEE 802.11b using single cell connection can be six with 0% packet-loss and more than six concurrent calls will cause some amount of packet-loss.

In [24] authors have set up a wireless network using IEEE 802.11n in order to identify the performance of real-time applications such as SIP (Session Initiation Protocol) and RTP (Real-Time Transport Protocol) protocols. The network test-bed was based on real equipment including, 3 wireless routers and 3 workstations. Three routers were placed with distance of 5 meter and 2 workstations were installed at both ends of the network to act as voice caller and voice receiver while 3rd workstation acted as SIP server at first end of the network. Overall it was concluded that different voice codecs have different performance such as G.711u used more bandwidth due to its higher bitrate and lesser compression method.

A study in [25] focused on the quality of VoIP traffic over WLAN using various techniques and identified a solution for better throughput and lesser delay. The wireless network was set up based on ad-hoc mode and multiple routing protocols were tested such as AODV (Ad-Hoc on demand routing vector), DSR (Dynamic Source Routing), OLSR (Optimized Link State Routing) and GRP (Geographic routing protocol). The network simulation tool known as OPNET was utilized to simulate different scenarios using IEEE 802.11g standard and G.711 voice codec. Thus, each routing protocol was implemented one by one and numbers of node were increased consistently. The results obtained indicated that AODV and OLSR protocols performed much better within range of 8 workstations. However, OLSR showed the ability to support

VoIP traffic as it provided the higher throughput and lesser delay comparing to other protocols tested.

IPv6 was designed to cover the shortage of IP addresses and to provide better services than IPv4. In [26] a study was carried out in order to identify the impact of WLAN on VoIP using IPv6. Hence, a new method was introduced using IEEE 802.11e standard to improve the quality of VoIP traffic for future networks. Comparison between VoIP traffic over IPv4 and IPv6 was conducted using multiple voice codec [27]. Multiple operating systems were involved and different parameters were measured such as RTT, jitter and packet-loss. The experiment was set to generate 10 concurrent voice calls using both versions of IP (one at time). The outcome pointed out that overall IPv4 marginally outperformed the IPv6.

VoIP on IPv6 infrastructure was established in order to measure the performance of VoIP calls, covering parameters such as jitter, delay and packet-loss. Soft-phones and IP phones were used to generate voice calls between users and 10 calls test were generated. Results observed for packet-loss, showed that 2 out of 10 calls provided packet-loss [19].

Performance of VoIP over IEEE 802.11a wireless network was investigated by researchers in [17, 18]. The impact of retransmission was estimated and the discussions conducted over channel errors on VoIP.

III. Experimental Test Bed

In this experimental research, two types of network test-beds were implemented using various configurations. Both test-beds included two workstations and were based on peer to peer environment. First network test-bed was wirelessly connected via physical AP (ph-AP) while second test-bed was connected using virtual AP (v-AP). Four different types of wireless NIC cards were installed on each workstation (one at a time). To secure the wireless connections we used WPA 2 personal security protocol during all tests. Two different internet protocol versions were established (IPv4 and IPv6) on both network test-beds one at time. The platform installed on all these workstations was Windows 7 as it supports virtual access point. A network test-bed diagram is shown below and illustrates the structure of our test-bed.

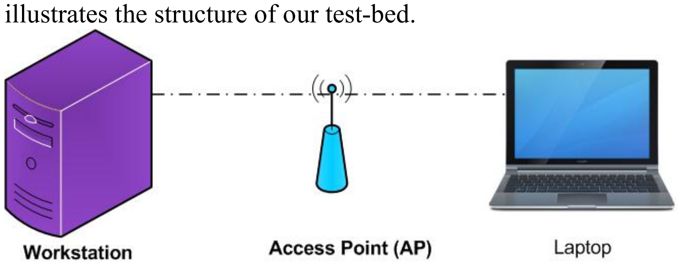

Figure 2: Network test-bed

Figure 2 shows our proposed network design: we created four WLAN scenarios such as WLAN-1, WLAN-2, WLAN-3 and WLAN-4. WLAN-1 includes Realtek wireless NIC cards installed on both machines while WLAN-2 includes Realtek and Intel(R) Centrino NIC wireless cards installed on each machine. WLAN-3 involves TP-Link and Intel(R) Centrino NIC wireless cards installed on each machine whereas WLAN-4 involves TP-Link and Realtek wireless NIC cards on each machine.

The hardware used in this experiment contains two computers (a workstation and a laptop). Workstation hardware included Intel quad core with 4.00GB RAM while a laptop was based on an Intel® Core™ 2 Duo with 6.00 GB RAM. The laptop had (Intel(R) Centrino(R) Advanced-N 6230) IEEE 802.11 NIC card built-in whereas a workstation had (TP-Link TL-WN951N) IEEE 802.11 NIC card installed with PCI slot. Two additional (Realtek RTL8191SU) IEEE 802.11 NIC cards with support of USB 2.0 were also involved. These NIC cards were turned into IEEE 802.11g mode which is capable of supporting 54Mbps bandwidth capacity (theoretically). A D-Link (wireless N750 dual band router with 3 antennas) access point was also used in this network setup.

IV. Internet Traffic Measureing Tool

Distributed Internet traffic generator (D-ITG) [8] is a network traffic measurement simulation tool that was selected to simulate VoIP traffic over WLAN. This tool is capable of generating and measuring different types of Internet traffic such as data, voice and multimedia. It also works across multiple operating systems and supports both version of IP (IPv4 and IPv6). The D-ITG uses fixed structure of frame sizes and packets per second for each of the VoIP codec as illustrated in Table 2 below:

Table 2: VoIP Codecs Specification [8]

Codecs	Samples	Frame size	Packets (per sec)
G.711.1	1	80	100
G.723.1	1	30	26
G.729.2	2	10	50

Selection of VoIP codec was determined based on the aim we had for this research. We selected a codec which is highly impacted by the WLAN to demonstrate the best benefit from our proposed method to improve the QoS. Our literature showed [9, 27] that G.711.1 is a voice codec among others which is highly impacted by WLAN.

V. Experimental Design

D-ITG command mode platforms were installed on both machines. Hence, D-ITG sender application was configured on a first machine to broadcast VoIP streams whereas second machine had D-ITG receiver application configured. D-ITG tool [8] was used to generate and measure the simulated traffic. Twenty concurrent VoIP traffic flows were simulated for each voice codec (G.711.1, G.723.1 & G.729.2) and each flow was equivalent to a VoIP call. The parameters considered include delay, jitter and packet-loss and 10 repeated tests runs were averaged. RTT (Round-Trip-Time) tests were measured for all experiments with less than 512kbps of background traffic.

VI. EXPERIMENTAL RESULTS

A. Experiment 1: Voice over WLAN (VoWLAN)

For the VoIP tests, we have considered two different IP protocols to identify the impact of the current IP protocol (IPv4) performance and the future choice of protocol (IPv6). We designed and configured four different WLAN scenarios such as WLAN-1, WLAN-2, WLAN-3 and WLAN-4. RTP with G.711.1, G.723.1 and G.729.2 voice codecs were selected with no VAD (voice activity detection). In this graph we present the VoIP test results for IPv4 and IPv6 as well as our designed scheme for WLAN.

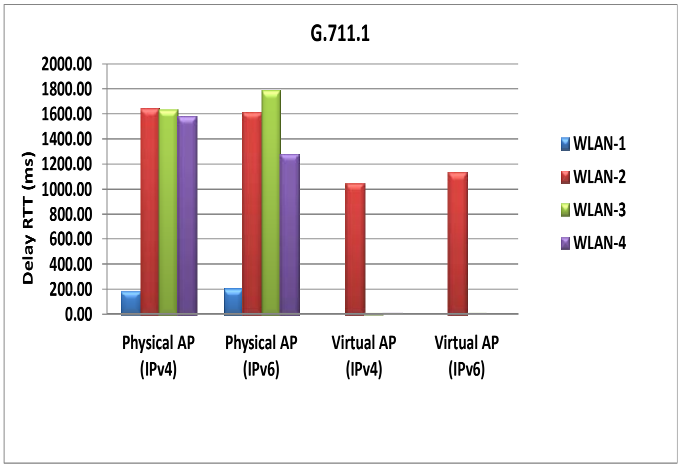

Figure 3: RTT comparison of VoIP traffic (G.711.1) over IPv4 and IPv6 using four WLAN scenarios

RTT results (in Fig.3) for G.711.1 codec on ph-AP using IPv4 over WLAN-2 showed that it provided highest amount of delay at approximately 1600ms while WLAN-1 outperformed its counterparts (WLAN-2, 3 & 4) as it produced approximately 1400ms lesser delay than WLAN-2, 3 & 4. The outcome for ph-AP using IPv6 showed that WLAN-3 provided highest amount of delay ~1800ms while WLAN-2 had second highest at ~1600ms. Once again WLAN-1 produced least amount of delay at ~200ms while WLAN-4 had second lowest amount of delay at ~1300ms. Comparison between IPv4 and IP6 using ph-AP indicated that WLAN-3 performed better on IPv4 as it provided ~180ms lesser delay than IPv6 while WLAN-4 performed better on IPv6 as it produced ~300ms lesser delay than IPv4. WLAN-1 and WLAN-2 performed similarly on both versions of IP (IPv4 and IPv6).

Comparison between ph-AP and v-AP using both versions of IP showed that v-AP outperformed its counterpart using four WLAN scenarios (WLAN-1, 2, 3 & 4). V-AP reduced RTT to minimum level for WLAN-1, 3 & 4 at approximately lesser than 10ms while WLAN-2 still had ~1100ms of delay for IPv6 and ~1050ms for IPv4. Overall it can be drawn that usage of v-AP can improve the delay performance for VoIP in most case scenarios.

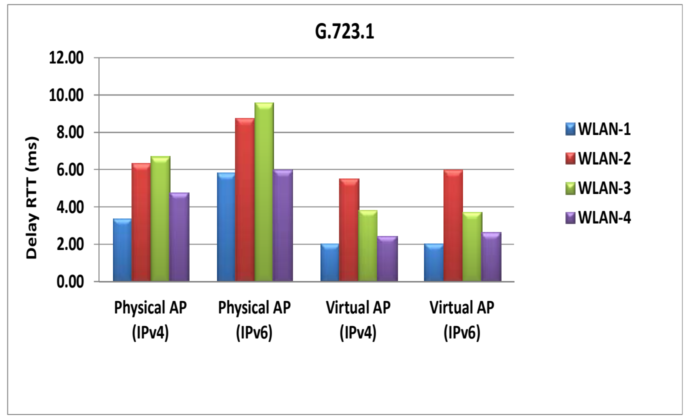

Figure 4: RTT comparison of VoIP traffic (G.723.1) over IPv4 and IPv6 using four WLAN scenarios

The results for G.723.1 codec on ph-AP using IPv4 indicated that WLAN-3 had highest amount of RTT at ~6.5ms while WLAN-2 had slightly lesser than WLAN-3 ~6.2ms. Least amount of RTT was observed on WLAN-1 using IPv4 at ~3.2ms while WLAN-4 had second lower RTT at ~5ms. The outcome for ph-AP using IPv6 showed that WLAN-3 provided highest amount of delay ~9.5ms whereas WLAN-2 had second highest RTT at ~9ms. WLAN-1 and WLAN-4 produced similar amount of delay at ~6ms which is 3.5ms lesser than WLAN-3. Comparison between IPv4 and IP6 using ph-AP was measured and outcome showed that IPv4 outperformed its counterpart using four WLAN scenarios. WLAN-1 using IPv4 provided ~2ms lesser delay than WLAN-1 using IPv6 while WLAN-3 using IPv4 produced ~3ms lesser delay than WLAN-3 using IPv6. Overall IPv4 marginally performed better than IPv6 over ph-AP.

Comparison between IPv4 and IPv6 over v-AP indicated that both versions of IP provided similar amount of delay for all four WLAN scenarios. However, comparison between v-AP and ph-AP showed that v-AP using IPv4 and IPv6 outperformed the ph-AP for all four WLAN scenarios. WLAN-3 using IPv4 on v-AP produced slightly lesser delay at ~1ms than IPv4 on ph-AP whereas WLAN-3 using IPv6 on v-AP provided ~5ms lesser delay than IPv6 on ph-AP.

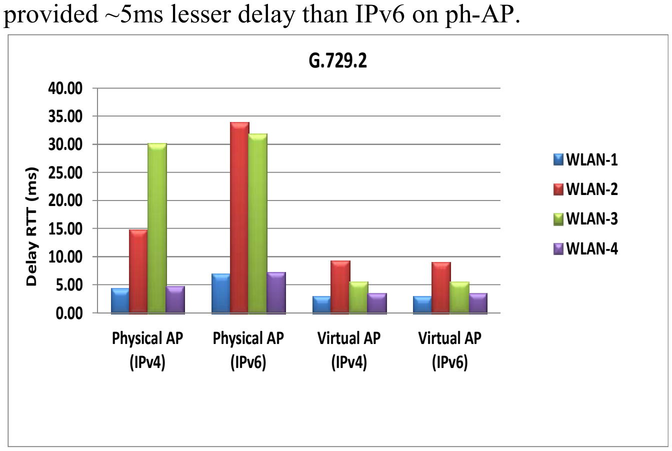

Figure 5: RTT comparison of VoIP traffic (G.729.2) over IPv4 and IPv6 using four WLAN scenarios

RTT results shown above (in Fig 5) indicated that ph-AP using IPv4 over WLAN-3 provided highest amount of delay at ~30ms whereas WLAN-2 provided ~15ms lesser delay than WLAN-3. WLAN-1 on ph-AP using IPv4 provided least amount of delay at ~4.5ms while WLAN-4 had similar amount of delay as WLAN-1. The outcome for ph-AP using IPv6 indicated that WLAN-2 provided highest amount of delay ~34ms whereas WLAN-3 had second highest at ~32ms. WLAN-1 using IPv6 on ph-AP had least amount of delay at ~7ms while WLAN-4 had second lowest amount of delay at ~8ms. Comparison between IPv4 and IP6 using ph-AP showed that WLAN-2 performed much better on IPv4 as it provided ~19ms lesser delay than IPv6 while WLAN-3 performed lightly better on IPv4 as it produced ~2ms lesser delay than IPv6. Overall ph-AP using IPv4 outperformed IPv6 using all four WLAN scenarios.

Comparison between v-AP and ph-AP using both versions of IP indicated that v-AP outperformed its counterpart using four WLAN scenarios (WLAN-1, 2, 3 & 4). V-AP using IPv4 reduced delay to minimum level especially for WLAN-3 as it produced ~25ms lesser delay than IPv4 on ph-AP. V-AP using IPv6 provided ~23ms lesser delay than IPv6 on ph-AP. Overall it can be drawn that usage of v-AP using IPv4 and IPv6 over four WLAN scenarios can provide better performance.

B. Experiment 2: VoWLAN(Jitter)

Experiment 2 emphasizes on VoWLAN results for jitter using ph-AP and v-AP under four WLAN scenarios.

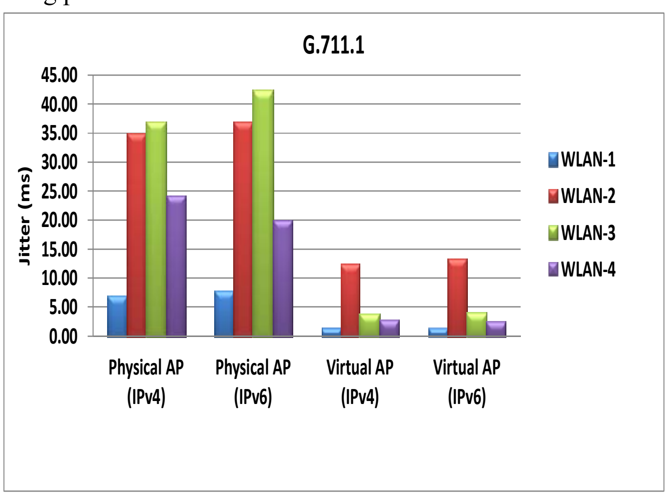

Figure 6: Jitter comparison of VoIP traffic (G.711.1) over IPv4 and IPv6 using four WLAN scenarios

The G.711.1 codec results for jitter showed that ph-AP using IPv4 over WLAN-1 performed better than WLAN-2, 3 & 4. Comparison between IPv4 and IP6 using ph-AP indicated that IPv4 performed marginally better than IPv6 over WLAN-1, 2 & 3 AP whereas WLAN-4 using IPv6 performed slightly better than IPv4 over ph-AP.

Comparison between ph-AP and v-AP using both versions of IP showed that v-AP outperformed its counterpart using four WLAN scenarios (WLAN-1, 2, 3 & 4). V-AP reduced RTT to minimum level for WLAN-1, 3 & 4 at approximately lesser than 3ms while WLAN-2 still had ~12ms of delay for IPv4 and ~13ms for IPv6. Overall it can be drawn that usage of v-AP can improve the delay performance for VoIP in most case scenarios.

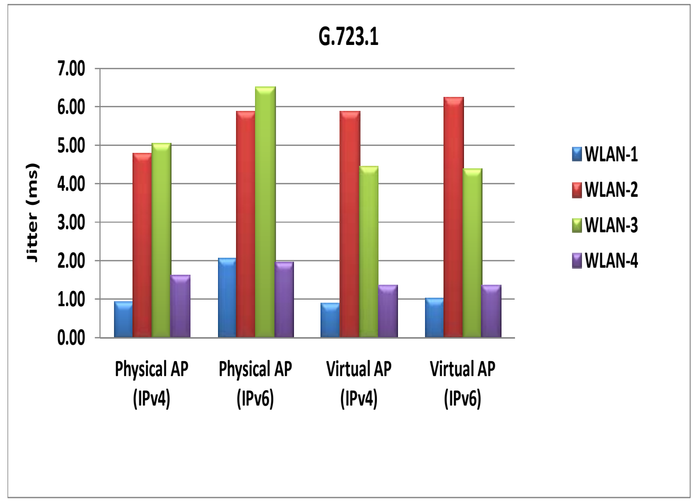

Figure 7: Jitter comparison of VoIP traffic (G.723.1) over IPv4 and IPv6 using four WLAN scenarios

The results for jitter shown in Fig.7 above indicated that ph-AP using IPv4 over four WLAN scenarios marginally performed better than its counterpart (IPv6). WLAN-2 & 3 had higher jitter than WLAN-1 & 4 on ph-AP using both (IPv4 & IPv6). Comparison between IPv4 and IPv6 using v-AP showed that they both performed similar results across all four WLAN scenarios. However, comparison between ph-AP and v-AP using IPv4 over WLAN-2 indicated that ph-AP using IPv4 over WLAN-2 performed slightly better than WLAN-2 on v-AP. Overall results observed for G.23.1 codec showed different results than other codecs tested in these experiments. It can be seen that v-AP provided similar results as ph-AP for IPv4 and IPv6 using all four WLAN scenarios. Furthermore discussions will be highlighted in more depth in discussion section.

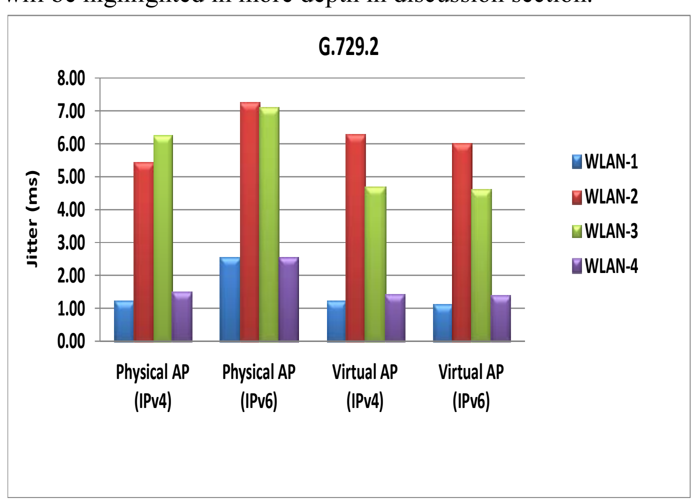

Figure 8: Jitter comparison of VoIP traffic (G.729.2) over IPv4 and IPv6 using four WLAN scenarios

The outcome for G.729.2 codec shown in Fig.8 above indicated that ph-AP using IPv4 performed slightly better that its counterpart (IPv6) under all four WLAN scenarios. WLAN-1 using IPv4 on both (ph-AP & v-AP) performed better than all three WLAN scenarios (WLAN-2, 3 & 4). However, WLAN-1 using IPv6 on ph-AP showed that it performed almost same amount of jitter as WLAN-4 using IPv6 on ph-AP.
Comparison between IPv4 and IPv6 using v-AP showed that they both performed similar results across all four WLAN scenarios. However, comparison between ph-AP and v-AP using IPv6 over WLAN-1 indicated that v-AP using IPv6 over WLAN-1 performed marginally better.

C. *Experiment 3: VoWLAN(Packet loss)*

The outcome for packet loss shown in table 3 below indicated that usage of ph-AP produced some amount of packet loss for all three codecs tested under four WLAN scenarios. WLAN-4 using IPv4 produced ~21.76% packet loss whereas WLAN-4 using IPv6 provided ~8.51% lesser packet loss than IPv4 for G.711.1 codec. G.723.1 codec had 0% packet loss when it was tested over WLAN-4 using IPv4. However, it provided ~0.01% packet loss using IPv6. The results for G.729.2 codec showed that WLAN-3 provided highest amount of packet loss at ~0.02% using IPv4 while IPv6 also provided same amount of packet loss. More discussion will be covered in discussion section.

Table 3: VoIP results using IPv4 and IPv6 over physical-AP (%)

	Physical AP (IPv4)			Physical AP (IPv6)		
	G.711.1	G.723.1	G.729.2	G.711.1	G.723.1	G.729.2
WLAN-1	2.11	0.02	0.015	4.57	0.01	0.02
WLAN-2	2.34	0.03	0.005	2.12	0.03	0.05
WLAN-3	078	0.05	0.02	1.51	0.04	0.02
WLAN-4	21.76	0	0.005	13.25	0.01	0.01

Packet loss results for virtual AP are shown in table 4 below. The results indicated that usage of v-AP produced lesser amount of packet loss comparing to ph-AP for all three codecs tested under four WLAN scenarios. The G.711.1 codec over WLAN-2 using IPv4 produced ~0.59% packet loss whereas WLAN-2 using IPv6 provided ~0.2% lesser packet loss than IPv4. The outcome for G.723.1 codec had 0% packet loss using IPv6 over four WLAN scenarios. However, it produced ~0.03% packet loss using IPv4 over WLAN-4. The G.729.2 codec also had 0% packet loss across four scenarios using IPv4 and IPv6 except for WLAN-4, where ~0.01% packet loss was observed using IPv4 and IPv6. More discussion will be covered in discussion section.

Table 4: VoIP results using IPv4 and IPv6 over virtual-AP (%)

	Virtual AP (IPv4)			Virtual AP (IPv6)		
	G.711.1	G.723.1	G.729.2	G.711.1	G.723.1	G.729.2
WLAN-1	0	0	0	0	0	0
WLAN-2	0.59	0	0	0.39	0	0
WLAN-3	0	0	0	0	0	0
WLAN-4	0.04	0.03	0.01	0.01	0	0.01

VII. DISCUSSION

In this study we evaluated the performance of VoIP over WLAN using IPv4 and IPv6 under four different scenarios. We also proposed a scheme to enhance QoS for real-time application of VoIP over WLAN. Our four scenarios were tested on two different network test-beds and the outcome is discussed in the following:

VoIP over random wireless devices vs. identical wireless devices: In this scenario twenty concurrent calls were established on WLAN using three different voice codecs. Four different WLAN scenarios were created (WLAN-1, 2, 3 & 4) and each scenario included two random combinations of wireless NIC cards. The results showed that two different bands of NIC cards could degrade the quality of real-time application. WLAN-2 and WLAN-3 scenarios using G.711.1 codec clearly indicated that impact caused by different wireless devices was quite high at approximately 1600ms delay whereas VoIP delay tolerance level indicates that it should be below 200ms. However, same tested were repeated on WLAN-1 which was based on identical wireless NIC cards and the results showed significant improvement.

VoIP over physical AP vs. virtual AP: In this case scenario, VoIP traffic was tested over a wireless network using a physical AP and a virtual AP. The outcome showed that physical AP impact the quality of VoIP compared to virtual AP. Physical AP has its own advantages but when it comes to QoS for real-time application, our results indicated that virtual AP is capable of handling large amount of real-time traffic with minimum impact on QoS comparing to physical AP. One of the main advantages of physical AP is it can be placed anywhere without any additional device but it does have limited processing power. Virtual AP runs on a computer but has much more processing power as it uses computer resources. This allowed it to perform much better than physical AP.

VoIP over IPv4 vs. IPv6: Considering the shortage of IPv4 which is expected in near future. Therefore, we setup a pure IPv6 based network and run all those tests which were run on IPv4 to identify the performance four scenarios over IPv6. The outcome for (VoIP) delay stated that IPv4 marginally outperformed its counterpart (IPv6) while results for packet-loss showed that IPv6 slightly performed better than IPv4 in most cases. Overall, it can be drawn that IPv6 would produce similar results as IPv4 for real-time applications.

VIII. CONCLUSION

In this paper we have investigated the performance of real-time application of VoIP over WLAN. We proposed a scheme of implementing a WLAN to allow users to have improved QoS for real-time applications. Our proposed scheme was tested on physical AP and virtual AP and the results indicated that our four scenarios performed much better on virtual AP as they provided minimum packet loss for IPv4 and IPv6 comparing to physical AP. RTT delay results also showed much better performed by virtual AP comparing to physical AP. Finally it can be concluded that combination of different wireless devices degrade the quality of real-time applications. However, it was also observed that, in our experimental setup, usage of identical equipment from the same manufacturer provided significant improvement. Our virtual AP scheme also reduced the impact of different wireless devices for real-time applications. The combination of random wireless devices also affected the performance of IPv6. Plus higher real-time traffic is highly impact by these devices where as lower 2Mbps applications are still considered within packet-loss and latency tolerance level.

Our future research will emphasise on other real-time applications, such as video traffic and real-time sensory data, in addition to different types of VoIP and gaming traffics. We will also investigate different standards of WLAN using various network scenarios. Additional evaluations covering more recent operating systems e.g., Windows 8, Android, etc., and newer voice, video and multimedia protocols are other area of interests.

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