Comparative Study of VoIP over WiMax and WiFi

M. Atif Qureshi*[1], Arjumand Younus*[2], Muhammad Saeed[#3], Farhan Ahmed Sidiqui[#4], Nasir Touheed*[5], and M. Shahid Qureshi*[6]

*** Faculty of Computer Science, Institute of Business Administration
Karachi, Pakistan**

**[#] Department of Computer Science, Karachi University
Karachi, Pakistan**

Abstract

VoIP is a technology in great demand these days. Its interactive nature makes it very appealing for users and today it is one of the most dominant technologies for communication. With the growth over wireless networks the option to have voice communication over wireless has been considered - the choices are VoIP over WiFi or VoIP over WiMax. This paper studies and compares the two options and summarizes the results.

Keywords: *Packet loss, jitter, throughput, congestion window size, QoS.*

1. Introduction

Recently wireless technology has grown immensely in popularity and usage becoming a medium of choice for networks. The wireless communication revolution is bringing fundamental changes to data networking, telecommunication, and is making integrated networks a reality. By freeing the user from the cord, personal communications networks, wireless LAN's, mobile radio networks and cellular systems, harbor the promise of fully distributed mobile computing and communications, anytime, anywhere.

A similar trend is seen in the world of voice communication and now transmission of voice over wireless communication links is very common as is obvious from the huge adoption of mobile telephony around the world [1]. One example of a rapidly growing voice application is VoIP as can be evidenced from high success rates of applications like Skype [2]. Voice over Internet Protocol (VoIP) technology facilitates packet based IP networks to carry digitized voice, it uses Internet Protocol for transmission of voice as packets over IP networks [12] thereby dramatically improving bandwidth efficiency and facilitates creation of new services. VoIP has enabled service providers to offer telephony services along with traditional data services using the same IP infrastructure and this in turn leads to improvement of business models.

However one fundamental question that arises is: “Can we get good VoIP quality over wireless networks while at the same time maintaining its traditional role for data services?”

We have addressed this question in this study by doing measurement analysis of VoIP over both WiFi and WiMax networks. The approach adopted is based on simulation using the well-known networking research simulation tool ns2 [3]. We performed two experiments: one for the case of IEEE 802.11 and the other for the case of IEEE 802.16.

VoIP packets are sent in conjunction with TCP packets and the performance of network is analyzed through various characteristics such as jitter, packet losses, throughput and delay.

This paper is organized as follows: Section 2 discusses the issues that arise when using VoIP over wireless networks. Section 3 explains the simulation scenario. Section 4 presents measurement results and graphs along with explanation along with an explanation of the results.

2. Voice going Wireless

Voice is the method of choice for real time communications [4]. Voice is so important to human communications that we have constructed entire networks centered around voice, namely, the public switched telephone network (PSTN) [5] and the analog/digital cellular networks [6]. Computer networks were originally developed with data transmission in mind, but the needs of Internet users today are diverse; no longer is the need for transmitting only data traffic over the Internet but there is also need to make VoIP calls, play online games and watch streaming media. Indeed, voice over the Internet Protocol (VoIP) is growing rapidly and is expected to do so for the near future. A new and powerful development for data communications is the emergence of wireless local area networks (WLANs) in the embodiment of the 802.11 a, b, g standards [7, 8], collectively referred to as Wi-Fi [8]. Because of the proliferation and expected expansion of Wi- Fi networks, considerable attention is now being turned to voice over Wi-Fi, with some companies already offering proprietary networks, handsets, and solutions. However deployment of VoIP over WiFi poses some serious problems and concerns. This is the main reason why the shift is now towards WiMax.

In this paper we take up a comparative study based on measurement analysis of "simulated packet traces." The results are compared to see which option is more viable: VoIP over WiFi or VoIP over WiMax.

2.1 VoIP Issues on IEEE 802.11

Wireless Local Area Networks (WLANs) are increasingly making their way into residential, commercial, industrial and public areas. As VoIP applications flourish [2] voice will be a significant driver for widespread adoption and integration of WLAN. As such voice capacity of a WLAN, which is defined as the maximum number of voice connections that can be supported with satisfied quality, has been investigated in the literature [9, 10]. The capacity of G.711 VoIP using constant bit rate (CBR) model and a 10 ms packetization interval is 6 calls. The two main problems encountered when VoIP is used over WiFi are:

- The system capacity for voice can be quite low for WLAN.
- VoIP traffic and traditional data traffic such as Web traffic, emails etc. can mingle with each other thereby bringing down VoIP performance.

These problems exist mainly due to the following reasons:

a) There is large per-packet overhead imposed by WiFi for each VoIP packet – for both protocol headers and WiFi contention.
b) Design of 802.11 protocols is such that it allows clients to access the channel in a distributed manner which causes a contention for the network which is particularly evident in the case of VoIP due to the real-time nature of the traffic.

Hence in the case of VoIP over WLAN the perceived throughput and real throughput have a large difference. Even though it does seem as an attractive alternative to cellular wireless telephony it has several drawbacks as we shall further investigate in section 4 of this paper.

2.2 VoIP on IEEE 802.16

IEEE 802.16 [11] is the "de facto" standard for broadband wireless communication. It is considered as the missing link for the "last mile" connection in Wireless Metropolitan Area Networks (WMAN). It represents a serious alternative to the wired network, such as DSL and cable modem. Besides Quality of Service (QoS) support, the IEEE 802.16 standard is currently offering a nominal data rate up to 100 Mega Bit Per Second (Mbps), and a covering area around 50 kilometers. Thus, a deployment of multimedia services such as Voice over IP (VoIP), Video on Demand (VoD) and video conferencing is now possible, which will open new markets and business opportunities for vendors and service providers.
Concerning QoS support, the 802.16 standard proposes to classify, at the MAC layer, the applications according to their QoS service requirement (real time applications with stringent delay requirement, best effort applications with minimum guaranteed bandwidth) as well as their packet arrival pattern (fixed / variable data packets at periodic / aperiodic intervals). For this aim, the initial standard proposes four classes of traffic, and the 802.16e [11] amendment adds another class:

- Unsolicited grant service (UGS): supports Constant Bit Rate (CBR) services, such as T1/E1 emulation and VoIP without silence suppression.
- Real-time polling service (rtPS): supports real-time services with variable size data on a periodic basis, such as MPEG and VoIP with silence suppression.
- Extended rtPS : recently introduced by the 802.16e standard, it combines UGS and rtPS. That is, it guaranties periodic unsolicited grants, but the grantsize can be changed by request. It was speciallyintroduced to support VoIP traffics [11].

- Non Real-Time Polling service (nrtPS): supports non real-time services that require variable size data bursts on regular basis, such as File Transport Pro- tocol (FTP) service.
- Best effort (BE): for applications that do not requireQoS such as Hyper Text Transfer Protocol (HTTP).

Due to the above-mentioned QoS implementations on IEEE 802.16 VoIP performs better on WiMax as we shall see in the next section.

3. Experimental Setup

To investigate performance of VoIP with TCP on IEEE 802.11 and IEEE 802.16 simulations were undertaken using TCP flows along with CBR flows (defined on top of UDP flows). UDP was used for the VoIP data flow and the UDP packet properties were those of the G.711 codec [13].
Figure 1 shows the simulation setup in ns2. In this network both VoIP and TCP/IP data traffic will be used to test the network performance for VoIP.
The setup is composed of two wired nodes, three mobile nodes and a base station serving as the access point for the WiFi network in case of Experiment 1 and for the WiMax network in case of Experiment 2. In both the experiments the deployment of the network was kept the same but the TCP and VoIP flows were varied each time.
Also the number of flows was varied: the simulation part was done with ns2 whereas for analysis purposes the Linux utilities xgraph and gnuplot were used.

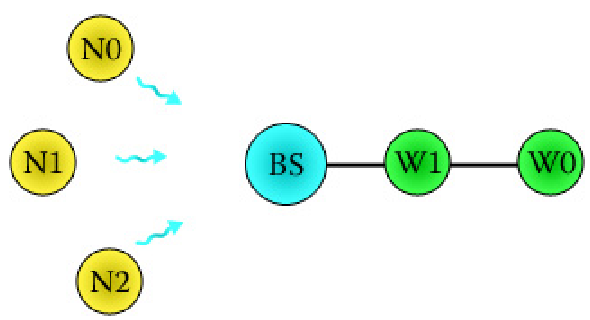


Fig. 1 Setup for Experiment

3.1 Experimental Details for Flows of TCP and VoIP

VoIP is basically CBR UDP: typical data rates and packet sizes can be obtained for voice codecs by doing a search for VoIP - typical data rates EXCLUSIVE of header overhead are from 5.3 to 64 kbps, depending on the implementation and application. Packet sizes are usually kept short to minimize latency.

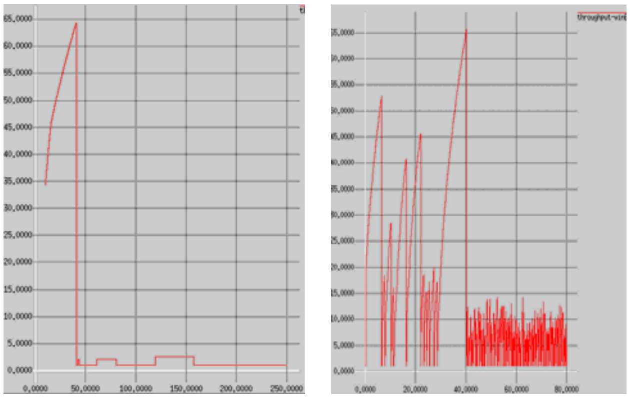
Hence in the ns2 simulation the VoIP packets have been modeled through CBR UDP with a data rate of 80 bytes and a delay of 20 milliseconds which is typical specification for G.711 codec [14].
In the case of the 802.11 scenario two TCP flows are set up: one from node N0 to wired node W0 (it is run from 5 seconds to end of simulation) and the other from wired node W1 to node N2 (it is run from 15 seconds to end of simulation). The VoIP packets are sent from node N0 to wired node W0 and from N2 to wired node W1. There are 16 VoIP flows instantiated simultaneously between N0 and W0 and their start time is 40 seconds, two of them are stopped at 100 second while remaining two at 120 seconds. Between N2 and W1 there are 4 simultaneous VoIP sessions with start times 100 seconds and ending times of the 4 are 120 seconds for first two, 140 seconds for third and 150 seconds for the last one.
In the case of 802.16 scenario the same example as the one provided by NS2 Simulator for IEEE 802.16 network [15] has been used and the topology for it has been shown in Figure 1. In the case of the 802.16 scenario three TCP flows are set up: one from node N0, node N2 and node N3 to wired node W1. Their start times are 0.1, 0.2 and 0.3 seconds and they stop when simulation ends; the VoIP packets are sent from node N0 to wired node W1. There are 8 VoIP flows instantiated simultaneously between N0 and W1 and their start time is 40 seconds out of which two are stopped at 60 seconds and remaining are allowed to run till the end of the simulation.

4. Experimental Results

This section presents the results for the two experiments. We plotted graphs for throughput, jitter and packet losses in both cases.

4.1 The 802.11 and 802.16 Results Compared

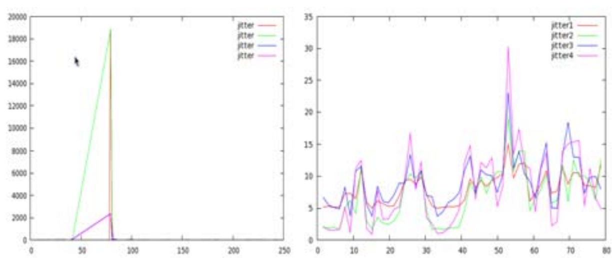
The through0put graph for both cases is shown in Figure 2. From the above graphs it is clear that VoIP over WiFi makes TCP capacity inefficient and as soon as VoIP flows are started the TCP congestion window drops and does not rise again until and unless the VoIP packet sending process drops. So this makes it clear that throughput of VoIP and TCP both are affected by deployment of VoIP over 802.11.
The first graph i.e. Figure 2a shows the scenario for the 802.11 networks and throughput that TCP achieves in presence of VoIP packets being transmitted.



(a) IEEE 802.11 (b) IEEE 802.16

Fig. 2 Throughout Window for TCP flows in 802.11 and 802.16 Networks

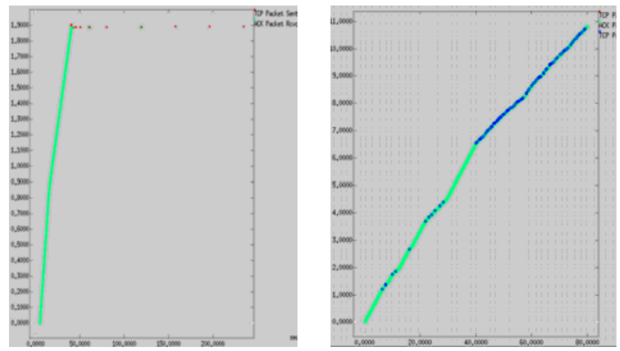
The graph shows that in presence of VoIP flows on WiFi the TCP capacity is marginally reduced and congestion window is affected badly. On the other hand in the case of WiMax networks the TCP does achieve an acceptable throughput hence demonstrating that WiMax is better suited for real-time services like VoIP.
The next graph in Figure 3 shows the jitter experienced by the TCP packets when VoIP flows and TCP flows exist simultaneously on a wireless link.



(a) IEEE 802.11 (b) IEEE 802.16

Fig. 3 Jitter for TCP flows in 802.11 and 802.16 Networks

Again this graph shows that VoIP existence on WLAN kills the TCP capacity of the network with very high jitter at times when VoIP packets co-exist and no jitter when there are no VoIP packets. However in the WiMax scenario VoIP packets do not make the network unsuitable for TCP thus proving the claim of WiMax community that "it is the ideal standard for both voice and data."
Figure 4 shows the graph for packet losses experienced by the TCP packets when VoIP flows and TCP flows exist simultaneously on a wireless link.
The red dots indicate sent packets, green dots are for received packets and blue dots mark the dropped packets over the network.
In the case of IEEE 802.11 networks we have almost no losses when VoIP is not being sent but as soon as we begin to send VoIP packets the congestion window gets halved due to packet drops at queue. Hence there is almost no further sending and receiving of packets and the network is unutilized by TCP since VoIP completely occupies it. Unlike that IEEE 802.16 networks although do show a packet loss but it is tolerable



(a) IEEE 802.11 (b) IEEE 802.16

Fig. 4 Packet Losses for TCP flows in 802.11 and 802.16 Networks

4.2 Other Characteristics

Although the graph has not been shown for the delay, it was however noted by analyzing the packet traces that as soon as CBR traffic was introduced into the network it took quite a long time for the TCP packets to arrive at the destination whilst at the same time VoIP quality suffered.

Moreover fairness was almost non-existent when number of flows was increased; the options of 4, 8 and 16 flows were tried for each case. In case of 16 flows the link containing VoIP traffic behaved as if it is down and due to that bandwidth of network was not equally shared.

4.3 Explanation of the Results

The results obtained and analyzed above were much expected due to the very nature of the two technologies of WiFi and WiMax.
There are three great problems inherent to the WLANs that can harm VoIP performance are:

- The inefficiency of the 802.11 MAC protocol.
- The signal instability caused by electromagnetic phenomena
- The competition for bandwidth usage between voice traffic and data traffic.

These problems not only make the performance of VoIP suffer over 802.11 but also render the network useless for data by choking TCP. On the other hand WiMax's better performance is attributed to its better QoS services.

WiMax is quite well suited to the promising VoIP applications.

4. Conclusions

All our findings complement the characteristics of both the networks and help in further establishing the fact that WiMax is better suited to VoIP than WiFi.

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