Study and Design of QoS-based VoIP over Wireless LAN

S. Korchi1, M. Boulmafl2, and A. Lakas3
1: AlAkhawayn University
Ifrane, Morocco
Email: {S.Korchi,M.Boulmal}@aui.ma
2: UAE University, Al-Ain, UAE

Abstract — Two of the major developments reshaping the telecommunications landscape are mobile wireless connectivity and the migration of voice telephone services to IP technology. These two ideas come together in networks that carry voice services over a wireless LAN (VoWLAN). The importance of usage of VoIP over wireless networks has been increasing, and a lot of research has been conducted to improve QoS and increase the capacity for VoIP traffic. In this paper, we investigate and study the performance of the VoWLAN network, by experimental approach using different scenarios with different codecs. We show that the quality drops significantly starting at 50 simultaneous calls, for the G729 codec. To be close to the reality, we performed the measurement with background traffic.

Keywords: VoIP, WLAN, QoS, design.

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 signalling protocols such as the Session Initiation Protocol (SIP) [1] and H.323 [2]. The use of packet technology introduces two major quality issues into voice service:

1. Voice Quality: The basic signal quality a user detects on a voice connection is a product of the voice coding technique that is used and the percentage of packets that fail to arrive at the receiver to be decoded. Packets can be lost two ways in a VoIP network:
   a) Packet networks can drop packets due to errors or buffer overflows. The impact of lost packets depends on the technique used to encode the voice.
   b) The RTP receive jitter buffer can drop packets if they arrive with a delay greater than the buffer can accommodate. So delivering a packet late is equivalent to not delivering it at all.

2. Transit Delay: Transit delay is the total delay the voice signal experiences as it travels through the network; this is also referred to as mouth-to-ear delay. A number of factors in the local and wide area network contribute to transit delay. These include voice coding/compression, packet generation, channel contention (in a WLAN), network transport/buffering, and jitter removal. The important thing to know is that once the one-way delay exceeds 150 msec, it will begin to affect the cadence of the conversation. Distance, router buffering, and WLAN contention are all contributing factors in end-to-end transit delay. Transit delay has been one of the major performance complaints we have seen in packet telephony.

The other timing issue is jitter or the variation in delay from packet to packet that is introduced by the dynamic buffering used in a packet network. Left untreated, jitter will render the voice unintelligible.

While these parameters are guaranteed using wired network, it’s a challenge to ensure any one of them in a wireless environment. The use of QoS specific MAC is not the miracle answer to these problems. Other parameters influence the quality of voice, and the experiment is a way of showing their influence.

In previous works, papers use simulations especially for 802.11g. The work on capacity measurement or performance evaluation is mainly focused on 802.11b networks as in [1] [2] [3]. But now the 802.11g is more and more used (especially through laptops), so a study of the capacities allowed by this standard is actual.

In this paper we focus on the experimental part in observing and analyzing how different parameters affect the quality of voice, that’s to say: codec and number of calls. The remainder of this paper is organized as follows: in section III we provide an overview of voice technologies explaining the characteristics of voice codecs and their QoS requirements. In section III, we describe the experiments we have carried out along with an analysis of the results obtained. We conclude by section IV.

II. TECHNOLOGY OVERVIEW

A) Voice codec and measurement

The quality of voice communication is greatly affected by the coding scheme used [4]. For PSTN, the PCM is standard codec used. Alternatively, there are four coding systems that could be used with IP voice.

1. G.726 (Formerly G. 723) Adaptive Differential Pulse Code Modulation (ADPCM): ADPCM was the first widely used compressed voice solution and it featured
To measure the quality of different voice coding systems and rate them on a scale of 1 to 5. A score of 5 is considered “perfect,” and 64 Kbps PCM produces a score of 4.4. A minimum score of 4.0 is recommended for business telephony, and 3.5 is considered marginally acceptable. The typical MOS value for a G.729A encoded connection is 4.0, and drops into the marginal range with one percent packet loss. The measurement can be computed for listening quality (LQ) and conversational quality (CQ).

To establish the listening quality, a set of phonetically balanced phrases like the Harvard Sentences (e.g., “The birch canoe slid on the smooth planks”) is played to a panel of listeners who are assigned a task to be conducted over the phone. Conversational quality is a more complex assessment as it addresses the overall call quality including listening quality, echo, and delay. The delay impact will vary based on the nature of the task (e.g., a business negotiation versus an informal chat).

B) Quality of Service used in WLAN

It is crucial to the success of deploying VoIP applications over WLAN to have the ability to support and provision QoS capabilities [5]. Furthermore, voice services inherently involve call control signalling that requires a high level of priority in order to meet the timing constraints of interfaces to external networks, such as the wireless cellular network or the public switched telephone network (PSTN).

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 [6]. The two following issues relate to time delay and jitter: (1) Signalling 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 signalling 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 [7]:

<table>
<thead>
<tr>
<th>Delay</th>
<th>Quality</th>
</tr>
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<tbody>
<tr>
<td>0 to 150 msec</td>
<td>Acceptable to most applications</td>
</tr>
<tr>
<td>150 to 400 msec</td>
<td>Acceptable for international connections</td>
</tr>
<tr>
<td>&gt; 400 msec</td>
<td>Acceptable for public network operation</td>
</tr>
</tbody>
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A new standard has been defined for Quality of Service. This standard defines the Enhanced Distributed Channel Access (EDCA)/Wi-Fi Multimedia (WMM). This is a contention-based access mechanism with four priority levels or access categories that can provide voice users with preferred access to the shared channel. The Wi-Fi Alliance identifies products that have tested in compliance with the EDCA option as Wi-Fi Multimedia (WMM) certified.

In EDCA/WMM, user priorities are implemented by enhancing two mechanisms from the original WLAN access protocol: the inter-frame spacing (IFS) and the back-off counter. In a traditional WLAN, a station wishing to send a frame must wait for an interval called the DCF Inter-Frame Spacing (DIFS). All users have the same access priority and drop into the same range of back-off values to address collisions and busy channel conditions (i.e., CWmin and CWmax). EDCA defines different waiting intervals and back-off ranges for each of four access categories, and shorter intervals are assigned to higher priority traffic.
The four EDCA/WMM defined priority levels or access categories are designated as follows:
- Access Category (AC) 1: Voice.
- Access Category (AC) 2: Video.
- Access Category (AC) 3: Best effort data (identical to current DCF devices).
- Access Category (AC) 4: Background data.

III. VoWlan EXPERIMENTS and result:

A) Experiment Setup

The setup of the two first experiments consists of closed network containing one Netgear WG302 ProSafe 802.11g Wireless Access Point supporting IEEE 802.11g and QoS mechanism, and two Windows based laptops, each equipped with IEEE 802.11g LAN card and a Windows based server wired to the access point. The setup is illustrated in the diagram depicted in Figure 1. The LanTraffic [8] software is used as traffic generator and analyzer which have two program components: one running on the server, and one running on one of the client laptop. The server program generates traffic destined to the client laptop. In these experiments, we have focused on the traffic throughput and the delay jitter. We also analyzed the correlation between these parameters. In addition, we have used WinSIP [9] and WinEyeQ (from Touchstone Inc.) [10]. WinSIP is used to generate VoIP traffic, and configure it regarding codec used, duration and error handling. WinEyeQ is used to monitor the traffic between the WinSIP clients. This software does a lot of measurement to have an idea about the voice traffic and its quality.

B) Results

The first experiment, we used two major codecs used commonly in VoIP: G729 and G723. These codecs allows more calls sessions, as they consume less bandwidth. At the opposite of the G711, these two codecs are more robust to voice degradation, because even with audio packet loss, the quality stays acceptable (greater than 3). In this experiment, we try to find the relationship between jitter variations, voice quality and the number of calls. To be more realistic, we added gradually background traffic to see how much it will affect the voice traffic.

Figure 3: Jitter per number of calls without QoS

Figure 3 shows that Jitter variation is proportional to the number of calls, but it’s more controlled in a QoS provided environment, as it’s visible in the Figure 4. The Jitter reaches 5 ms with AP configured without the QoS mechanism. With QoS mechanism enabled we measured only 2 ms with 30 calls. This assessment remains true independently of the amount of background traffic (but the jitter limits change).

Figure 4: Jitter per number of calls with QoS

Because we need to perform a handover, we installed a network with two APs and the server in the same network segment, so that the change on BSID does not affect the IP traffic. Concerning the wireless cards, we used two types: a standard wireless card of the laptop, and then a Cisco Aironet card that implements an algorithm to handle the handover.

Figure 5: MOS per number of calls without QoS

Figure 5 shows that the quality of voice, measured here with the MOS metric, drops significantly after a given number of calls, but this decrease is moderated when using QoS. With
50 calls the measured MOS is equal to 1 which a very poor quality, it means the voice is unintelligible. This remark is true for G729, the most robust codec for voice over IP. We point out that like G729, other codecs see their quality drops but with fewer voice calls.

Figure 6 depict the MOS with enabling the QoS mechanism in the AP. The results show that the MOS remind constant and equal to 4 which a good value. This is due to the priority given to the voice packet in the laptops cards. This conclusion needs is very specific because we have only one user of the wireless channel. Further simulations will give a more generalized conclusion to this conclusion.

Figure 6 : MOS per number of calls with QoS

Figure 7 gives the results of the MOS versus different type of codecs. It’s clear that only two codecs reach the fair quality (G729 and G723). The codec G729 has the advantage of having a less average delay for a communication than the G723 codec. This confirms that G729 provide an optimum quality for VoIP even if it consumes more bandwidth than G723.

Figure 7 : MOS versus codec

The packet loss is very degrading to the quality of voice, but does not happen until the number of calls reaches high values (beginning from 50, depending on the codec used).

Figure 8 : Comparative Jitter values with RTS/CTS enabled or disabled.

The main purpose of the second experiment is to see the effect of RTS/CTS on the network (Figure 8 and Figure 9). So by having 30 calls between clients and servers, and generating background traffic, we measured the jitter and the MOS. From it, we conclude that the RTS/CTS mechanism have a negative effect on the jitter and quality of voice. Even when using the QoS, the result obtained without the RTS/CTS mechanism where a little better. This is due to the little number of clients (2 clients), which is make the result without the contention mechanism unaffected by a lot of collision. The simulation will give more generalized result for this case.

In this figure, we show the variation of MOS with or without RTS/CTS mechanism. The value of MOS drops when the RTS mechanism is enabled, without affecting considerably the quality of voice.

Figure 9 : Comparative MOS values with RTS/CTS enabled or disabled.

The objective of the third experiment is to see the effects of handover over the feasibility of a VoIP call. For this, we used two different wireless cards: a laptop card with default configuration, and a card with handover capabilities for example Cisco Aironet card. The results showed that the algorithm implemented in the Cisco card made the handover safe for the communication even if the number of handover was greater. On the other hand, the pc card was inadequate as the communication has to be interrupted and reinitialized, which make call with a handover using this card unfeasible.

Figure 10 : the evolution of jitter and throughput during a handover with a standard card.

As the previous figure shows that during a handover using a standard card, the jitter reaches values near 1200 ms. We see also that the throughput drops to values near 0. In that case, the number of packet loss is large, and this means that there is no voice communication. In the case of my experiment, the timeout has been reached and the calls had to be reinitialized.
IV. CONCLUSION

In this paper, we have presented a study on the quality of VoIP over WLAN. The study was oriented towards the assessment of the variation of the MOS and the packet delay jitter. The result showed that for two PCs sharing the media, the quality drops significantly beginning from 50 calls, for the G729 codec, the most appropriate codec for VoIP. So implementing a VoIP calls over wireless network suffers from a lot of problems due to each of the technologies. The QoS mechanism increased the quality, but the improvements weren’t much distanced from the ones without QoS. These remarks could be verified and generalized to a more complex network through simulations. This is my future work to complete the purpose of this paper, in addition to design a quality of service mechanism that could improve the quality of VoWLAN.

REFERENCES


