VoIP over WLAN 802.11b simulations for infrastructure and ad-hoc networks

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Abstract: Voice IP (VoIP) over wireless local area network (WLAN) is a topic of great interest in the research arena. To investigate what are the performance limitations in establishing VoIP calls over WLAN networks we performed some simulations. We show in this paper these simulation results in ad-hoc and infrastructure WLAN networks, working under ideal and real conditions. Measures and graphs of packet delay, packet jitter and lost packet rate show the performance limitations when VoIP runs over WLAN. The basic question posed was: how many VoIP calls can be established over a WLAN 802.11b network?

1 Introduction

Trying to establish a clear objective in our research from the beginning, we established the principal objective as a question: How many VoIP calls can be established over a 802.11b WLAN for ad-hoc and infrastructure architectures?. To answer this we measured three quantities which are the most important parameters involved in real-time voice communications, assuming that raw data throughput is not a problem. The first parameter is packet delay, which is a measure of the time taken by a packet to travel from an originating node to a destination node. The VoIP maximum delay recommended by ITU-T G.114 is 150 miliseconds [1]. The second parameter is packet jitter, which is a measure of delay variability. For this purpose we measured the time between packet arrivals. We checked the time variability between arrivals, because for jitter the average is not important but the variance is. The last parameter is lost packet rate. This is a measure of how many packets are dropped in a particular conversation. The results are given as a percentage, which will give some clue as to the impact on quality.

2 The Simulated Scenarios

2.1 VoIP Traffic

Our choice was to use a 64 Kbps half-duplex UDP flow. To generate a 64 Kbps flow, 64 bytes packets were sent every 8 miliseconds. This value, 8 miliseconds, is important for the jitter, because since we measure the time difference between arrivals, if the environment is not saturated and the calls are working properly the jitter value should be oscillating around 8 miliseconds.

2.2 Scenarios Simulated

Five scenarios were simulated. Three of them under an infrastructure mode architecture and two using an ad-hoc mode network. Some of them are ideal scenarios by which we mean that nodes in these simulations were very close together, something that may not happen in a real scenario. The reason to simulate an ideal scenario was to obtain a baseline maximum number of calls value and which we could then compare with some more realistic scenarios.

The simulated scenarios were: Infrastructure mode ideal scenario (AP packet queue length=50 packets. 20 nodes plus AP); Infrastructure mode ideal scenario (AP packet queue length=100 packets. 20 nodes plus AP);Infrastructure real scenario(20 nodes plus AP); Ad-hoc mode ideal scenario(31 nodes); Ad-hoc mode real scenario(31 nodes).

The key point in the infrastructure mode scenarios is the AP. When more conversations are added the AP has to schedule more packet transmissions, hence its packet queue will lengthen. These packets which are waiting to be forwarded will be delayed. The AP queue has a limited capacity and when the queue is full, if a new one is received it will be dropped.
In the ad-hoc case, a routing protocol has to establish routes among nodes in order to establish end to end user conversations. In this case, there will normally be intermediate nodes in a call. Depending on which nodes establish a VoIP call there will be more or less hops needed for a packet to arrive at the destination node. It is possible that some node in a conversation path will become saturated, because it could be a node involved in many conversations. In this case, all the conversations which use that node will be affected, their packets will be delayed and, if the node packet queue is full, some packets will be dropped.
This is the normal behaviour in an ad-hoc network. However, in our simulation there is a special ad-hoc scenario which presents ideal conditions (nodes very close). In this scenario there will not be any intermediate nodes in the conversation path, because the nodes are very close to each other, and only one hop is needed for a packet to reach its destination. Therefore, if there is high delay it is because the environment is saturated and many collisions are happening.

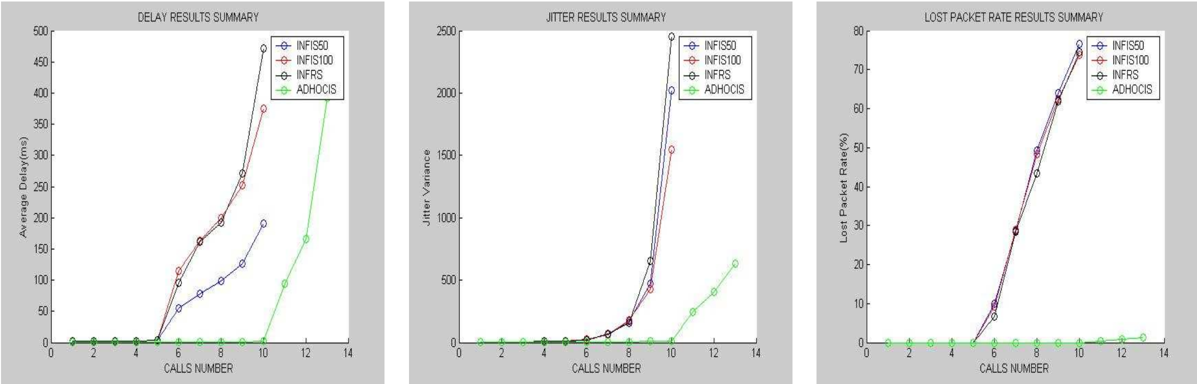
3 Results

The simulation results are reported in several figures. Figure 1(a) shows average packet delay vs number of calls running over a scenario. Figure 1(b) shows jitter variance vs number of calls. Figure 1(c) shows lost packet rate vs number of calls. These graphs show results for all scenarios except the ad-hoc real scenario.

The first scenario shows an abrupt change when the sixth call is added. The lost packet rate changes from 0% to 10%. This is an unacceptable value, because 10% of the information in a phone call cannot be lost, because the conversation would become not understandable. Also the delay results indicate bad quality, because the average delay is over 50 miliseconds. This means that the delay of many packets is over this value, whereas in the previous cases it has a very low value. The jitter's variance starts to increase at this point (sixth call) but not by too much, but it is a warning showing that something is working worse than in the previous cases. If even more calls are added, all the parameters get worse.

The second scenario results are very similar. We simulated this scenario because in the previous one we obtained an unacceptable lost packet rate when the sixth call was added. The AP drops packets when its queue is full. Trying to reduce this effect we doubled the AP's queue length. We expected to reduce the LPR whilst paying with a higher packet delay. Results obtained shows that when the sixth call is added the average delay is higher than in the first scenario, more or less the double, which means around 100 miliseconds of average delay. This is an unacceptable value because for much of the time the packet delay will be over 150 miliseconds to obtain an average of 100 miliseconds. However, a lower value of dropped packets does not appear. The result is an LPR of 9% instead of 10% with a 50 packet queue in the AP. The commentary for the jitter's variance is also the same than in the previous case. With the sixth call it starts to increase. Again, very bad results are obtained if 6 or more calls are running over this scenario.

The third scenario is a real infrastructure mode scenario in contrast to the previous ideal scenarios. In this case the nodes are located around the AP, as will occurs in many real cases. However, results in this case are quite similar to the previous ideal results. Again, when the



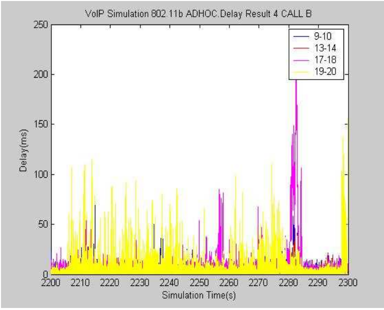
(a) Average packet delay (b) Variance packet jitter (c) Lost Packet Rate

Figure 1: Results

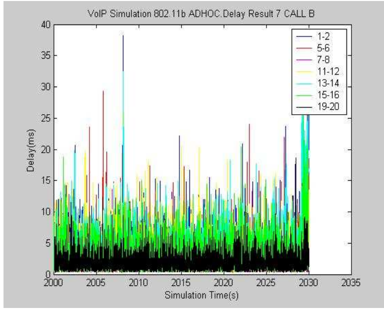
sixth call is added the call quality becomes unacceptable. Specifically, the average packet delay becomes around 100 miliseconds. The lost packet rate is slightly lower than in the previous cases, around 7% but this is still too much information being lost. The variance in jitter again presents the same bahviour than in the previous cases.

Results for the ad-hoc mode ideal scenario are reported in figures 1(a), 1(b) and 1(c). In this case the key step occurs when the eleventh call is added. Now, delay and jitter are the parameters which show unacceptable values. The average packet delay is around 100 miliseconds, and as we explained in some previous cases this is a bad value because, if the average is 100 ms, it is because many packets have a delay over 150 miliseconds which is not acceptable. The jitter's variance presents an abrupt jump between the tenth and eleventh calls. A high variability in the jitter shows bad quality in the communication, in this case very bad quality. The lost packet rate changes from 0% to 0.3%, which is an important change. Since it is an ideal scenario and packets need only one hop to arrive the destination node, if there are dropped packets it is because in the originating node queue there are 50 packets waiting to be sent, and a new one is generated and dropped because the queue is full. Therefore, this low lost packet rate explains why the delay is so high. A quick conclusion is that eleven or more calls cannot be established in an ad-hoc scenario with ideal conditions.

In the ad-hoc real scenario we cannot present a summary result, because the results in every simulation were different. It depends on which calls are running in the scenario and which nodes are involved in these calls. To explain this, figures 2(a) and 2(b), which are measuring the instantaneous delay of each packet in the scenario, are shown. In figure 2(b), seven calls are running properly over the scenario. It is easy to check that packet delay is always under 40 miliseconds and most of the time under 20 miliseconds, which is an acceptable value. However figure 2(a) shows the results with just 4 VoIP calls running with poor values for the delay, with a packet delay value over 50 miliseconds much of the time in 2 calls, and one of them with peaks which reach more than 150 miliseconds. If the jitter graphs were shown we could see a much more higher variability in the 4 calls simulation. Finally, in the 4 calls simulation there were dropped packets in one call, whereas there were not dropped packets in the case with 7 calls.
Other scenarios were simulated, and we found scenarios with 4 calls with good parameters' values and scenarios with 7 calls with unacceptable quality in the conversations.
We did not find a simulation with 8 calls working properly. We simulated many times with different combinations of calls, but no simulation presented good results.



(a) ad-hoc real scenario with 4 calls



(b) ad-hoc real scenario with 7 calls

Figure 2: Results ad-hoc real scenario

4 Conclusions

In this paper we have evaluated the performance of VoIP calls over 802.11b WLANs, in infrastructure and ad-hoc mode architectures. We have proposed and discussed several scenarios with different features.

We measured different parameters, involved in every real-time communication, such as packet delay, jitter and lost packet rate, and on the basis of them we have determined the maximum number of VoIP calls which can be supported by a WLAN.

The most important result obtained in all infrastructure mode scenarios simulated is that 5 VoIP calls can be established. When the sixth call is added, in the three infrastructure mode cases, the lost packet rate was not 0% anymore and the average delay increases abruptly, in one case up to 50 miliseconds, in the other two up to 100 miliseconds. Jitter's variance presented a slight increase. Worse results were obtained if more calls were added.

A similar conclusion could be made in the ad-hoc ideal scenario, when the nodes are very close together. In this case 10 calls can be established at the same time without problems. When the eleventh call was added, average delay and jitter variance grows until they reach unacceptable values. At this point a lost packet rate that was not 0% appeared. It was a low rate of 0.3% but with an important meaning because it expalins why the delay is so high.

There are no conclusions with values in an ad-hoc real scenario. The reason is because in this case every simulation made presented a different behaviour. In a real ad-hoc scenario it becomes very important which nodes are establishing a conversation and which are the intermediate nodes. Whether the calls can be supported is a function of node distribution and which nodes are involved in the calls.

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