Cross-layer measurements using voice in 802.11b networks

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1 Abstract

This paper reports on the suitability of 802.11b networks for real-time voice traffic. We have measured 802.11b VoIP parameters in an outdoor environment, inside an office and considered the effect of competing traffic. Additionally we have investigated the influence of the 802.11b protocol itself. We conclude that 802.11b can support real-time voice sessions for the scenarios included in this paper with the notable exception of a VoIP stream competing against multiple TCP sources in infrastructure mode.

2 Introduction

Fourth generation cellular networks are already being planned. Seamless mobility between network types is a popular thrust in networking research today. Nevertheless, currently deployed networks have not yet solved the problems of creating the impression of a single communication network for the average user, independent of the underlying technology. IEEE 802.11x and GSM are the most prevalent wireless networks today, and despite differences in their regulatory, technical, and usage scenarios, they *could* act as one voice network for the user. An 802.11 network can carry voice at the office, home, or at hotspot areas, while GSM can offer coverage where there is not adequate 802.11 coverage. This paper therefore addresses the well documented issues for real-time voice over 802.11b networks through extensive measurements.

3 Approach and method

Our method was to investigate the *application* perceived quality in different usage scenarios due to the environment's effect on VoIP quality. By environment we mean the physical context: the separation of the nodes, the intervening obstacles, and the effect of other users' traffic. We look at the quality variations of a single VoIP call in various circumstances: in point-to-point mode where clear line of sight is established, inside and outside an office environment, in the presence of competing TCP/UDP traffic, and with/out an access point. Where possible, we examine the contribution of the MAC layer and try to determine its influence on the application layer. Using information from different network layers simultaneously is often referred to as a cross-layer approach. We stop short of direct physical layer measurements, for example the effect of the signal strength on voice quality, although the signal strengths have been recorded for future study. Quality of service in this work is quantified by loss, delay, and jitter as seen at the application layer.

4 Cross-layer measurements

The basic configuration used for our experiments comprises one node that sends a flow of VoIP packets and a second note that acts as a receiver for the RTP stream. The configuration we used is shown in figure 1. In order to observe and capture the traffic in the air, we used two monitoring devices placed close to

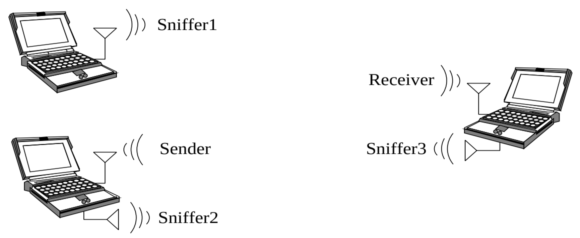

Figure 1: Cross-layer measurement testbed

the sender station[1]. One monitor was in the same sending station and one was physically separate. We studied the details of the traffic sent back from the receiver using a third monitor; they are mostly ACK frames. Ethereal and Sphone (an in-house VoIP tool [1]) were used to capture the MAC frames and VoIP packets respectively. During the experiments, the RTP traffic corresponded to a simulated call of 80 seconds generating 160 byte packets 20ms apart, no silence suppression or signaling were used. The default number of transmission attempts at the MAC layer is up to four, after which the frame is discarded by the link layer.

5 Goals

The goals are the use of cross layer information to:

- Understand the MAC layer contribution to the overall quality
- Provide an explanation of loss, delay, and jitter as seen at the application
- Help select appropriate data rates in particular situations

6 An example

Before addressing these goals, we will show a simple case with sender and receiver using ad hoc mode and separated by some meters; this illustrates the effect of retransmissions at the data link layer on delay and the limited RTP losses seen at the application layer for a single VoIP flow. Figure 2 shows these two quantities together. The x-axis shows ∼4000 packets and

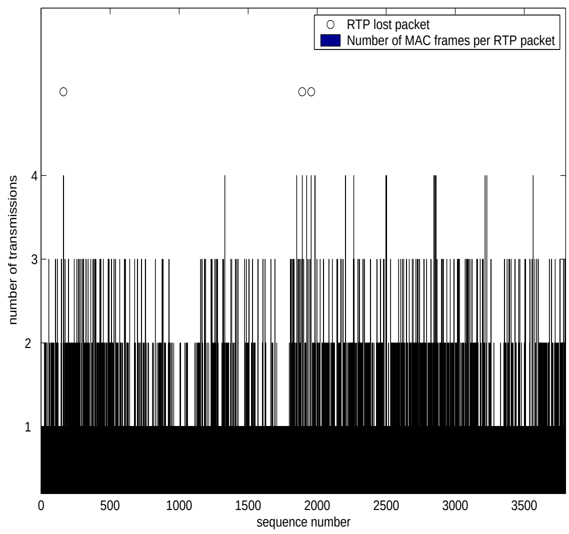

Figure 2: MAC transmissions vs. sequence no.

the y-axis indicates the number of transmissions recorded at the sender; as stated above this is always between one and four. Note that most of the frames are received successfully after the first transmission (shown as $y = 1$) with a small number of frames lost (indicated as circles). From the figure we can also see that most of the fourth transmissions were indeed successful. However, only by using the application layer information at the receiver, can it be established whether the fourth MAC retransmission was actually received (as the receiver's ACK may not always be received).

7 Results

7.1 Quality as a function of distance

We now examine the effect of distance between the sender and the receiver when outdoors.

[1]An experiment showed that the probability of both monitors losing a MAC frame was 0.04%.

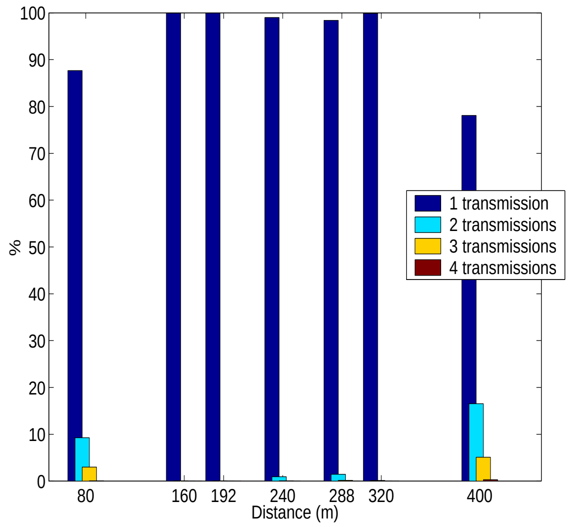

Figure 3: MAC transmissions vs. distance

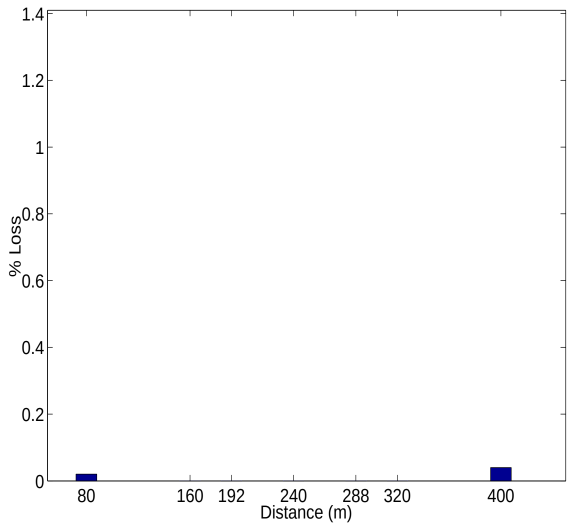

Figure 4: Losses vs. distance

Figure 3 shows a histogram of the percentage of MAC losses at seven different distances. At each distance we show the distribution of the number of transmissions necessary to transfer one voice packet. One can see that some retransmissions were necessary at 80 and 400 meters. We believe this is because of interference at the 80 meter range and possibly low signal to noise ratio at the 400 meter range. Figure 4 shows the application layer loss for the same distances with small loss percentages at the 80 and 400 meter distances. Note no significant competing traffic was observed on the channel. The delay and jitter values are also low (< 7ms), more details on the effect of the MAC layer on VoIP can be found in [3].

7.2 Office environment

We measured the number of MAC transmissions inside an office environment. The goal was to see the contribution of the walls, windows, and intervening obstacles typically found in this scenario. In this set of experiments we fixed the transmission rate and observed the effect of the environment and rate selection on the number of MAC transmissions. From figure 5 we can see that the default sending rate of 11Mbits/sec led to retransmissions and can even lead to losses. The important point is that in this environment reducing the rate is sufficient for the first transmission to be successfully received as can be seen from the 5.5Mbits/sec values. We can state through observations that the environment significantly affects the quality and in some situations even led to communication not being possible. More details of the application layer measurements can be found in [2]. A clear result is that reducing the bitrate immediately, rather than attempting retransmissions at a higher rate may alleviate loss, and reduces delay and jitter for a single frame.

7.3 Competing traffic

We now look at the effect of competing traffic on the quality of VoIP sessions. We consider both the cases of ad hoc mode (without an access point) and managed mode (with an access point). The VoIP nodes were in the same room with up to four nodes generating

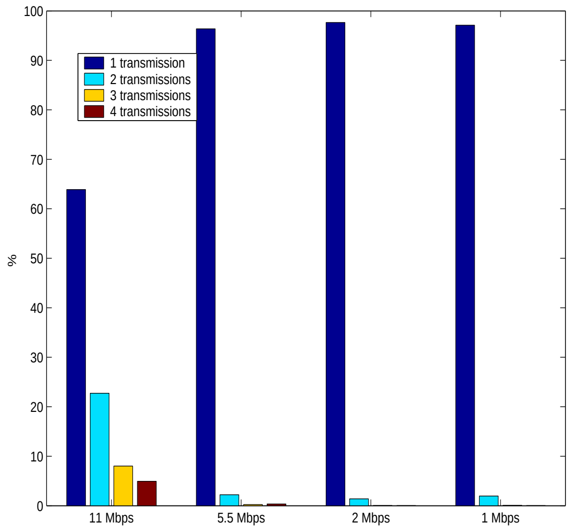

Figure 5: Successful transmissions vs. rate

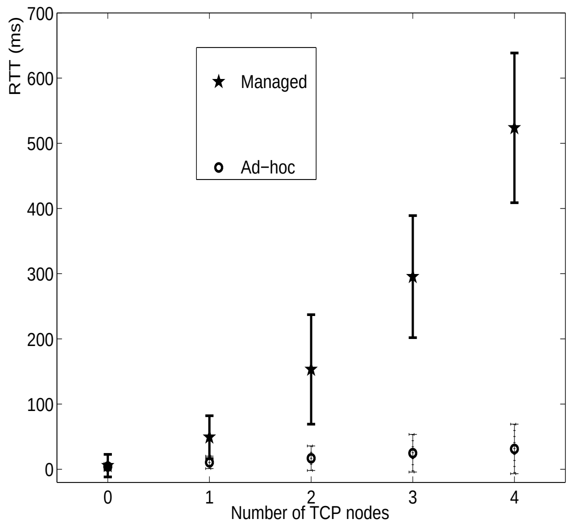

Figure 6: Ad hoc & managed mode RTTs

background traffic. Figure 6 shows the round-trip time for both IEEE 802.11 modes. Here, competing traffic was TCP based, generated using NTTCP. Note that the delay is higher in managed mode, but this is as expected due to the heavier load on the medium and the equal access mechanism, note in managed mode, the frames appear (at least) twice on the medium. This can lead to a round-trip time that would be unacceptable for VoIP users, exceeding the delay for good interactivity. We observed some losses (similarly jitter), but both were low and within the acceptable values for VoIP quality for the G.711 speech coding we used.

8 Conclusions & future work

The MAC layer behavior in *conjunction* with application layer performance is both useful and informative in estimating the quality of VoIP sessions. We have examined in two scenarios (outdoors and in an office) the effect of the MAC protocol when faced with different environmental conditions. We have found that radio interference and competing traffic pose threats to VoIP quality. In the referenced Masters theses we also show how intervening obstacles can degrade the voice quality significantly. Future work will look at the signal and noise levels, and combining the loss, delay, and jitter into a single quality value, as given by the ITU's E-model as well as simplified versions of it.

References

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