The effect of mobility on voice transmission capacity in mobile ad hoc networks *

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Abstract. *This paper analyzes the voice transmission capacity in mobile ad hoc networks by performing simulations related to end-to-end delay, jitter, and the number of consecutive losses. We evaluate the influence of mobility and nodes density on the number of voice transmitting sources. Results show that the network capacity for voice transmission in multihop mobile networks decreases, down to 40%, with mobility and network load variations. Moreover, long burst losses occur due to mobility, caused by the on demand routing mechanism and the lack of connectivity. The burst losses can achieve more than 2,000 consecutive packets in mobile multi hop networks, while in single hop networks this value is less than 90 consecutive losses.*

1. Introduction

Wireless communication plays an important role in computer networks due to its high flexibility and low implementation cost. Although wireless local area networks are becoming common place, the transmission of real-time traffic is still a great challenge.

Wireless communications can be infrastructured, where all communications take place through an access point like a cell phone network, or can operate on ad hoc mode, which is characterized by no infrastructure and nodes communicate directly to each other. The main advantages of ad hoc networks are flexibility, low cost, and robustness. Ad hoc networks can be easily set up, even in desert places and can endure to natural catastrophes and war. Therefore, ad hoc networks fit well where there is no infrastructure and it is too expensive to build it, or when local infrastructure is not reliable, as for instance, military operations in the enemy territory. On the other hand, in ad hoc networks each node must implement distributed medium access control mechanisms and deal with exposed and hidden terminal problems. These mechanisms add considerable complexity to nodes, especially in multihop networks, where they also act as routers. Besides, ad hoc networks must cope with other wireless problems, such as low transmission rate, high bit error rate (BER), and significant variations in physical medium conditions. This complexity makes transmission of real-time traffic a great challenge due to the Quality of Service requirements.

Some work have studied voice transmission in IEEE 802.11 networks. on infra-structured mode. Köpsel *et al.* [1] analyzed DCF (Distributed Coordination Function) and PCF (Point Coordination Function) mechanism with respect to the number of nodes transmitting voice traffic and proposed a hybrid mechanism using DCF and PCF modes. In order to improve network performance they also presented an optimal switching point from DCF to PCF mode. Wolisz *et al.* [2] presented an analysis of DCF and PCF considering the number of voice traffics and BER (Bit Error Rate). They showed that PCF performs better in high loaded networks and that increasing BER degrades network capacity. All these previous work consider the infra-structured mode.of the IEEE 802.11 networks. Results covering the ad hoc mode was not yet well explored.

This paper analyzes the effect of mobility on the capacity of voice transmission in ad hoc networks [3]. We also show the difference between the capacity of single hop networks and multi hop networks. Another issue addressed in this paper is the impact of network nodes density on the capacity of voice transmission. Therefore, we evaluate the behavior of QoS parameters, such as, loss rate, jitter, and consecutive losses, under different network conditions.

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The remainder of this paper is organized as follows. Section 2 briefly summarizes the main QoS parameters related to real time voice traffic. Simulation details and results are shown in Section 3. Section 4 presents our conclusions.

2. QoS in ad hoc networks

Real-time voice traffic has QoS requirements such as bounded end-to-end delay, maximum jitter, and limited loss rate. Voice traffic, different from data traffic, supports a limited packet loss rate and, moreover, is sensitive to the number of consecutive packet losses. Another important aspect is that the audio stream must be presented at the receiver with the same temporal relationship as it was captured. Therefore, jitter turns out to be an important QoS parameter, which is strongly related to synchronization and, consequently, to buffering at the receiver.

End-to-end delay consists of four basic components and plays an important role in interactivity loss. It includes coding/decoding delay, packet generation delay, propagation delay, and queuing delay. Table 1 presents some reference values of tolerance to delay recommended by the ITU-T [4].

Table 1: Tolerance to delay in voice communications.

Delay (ms)	Tolerance
less than 150	good interactivity
150-400	user can notice some loss of interactivity
over 400	Loss of interactivity

In this paper we consider these three QoS parameters in the analysis. At first, we simulated voice transmission in a single hop ad hoc network and then in a multihop network in order to compare the capacity in these networks. The capacity is given by the number of sources transmitting voice, according to the maximum loss rate previously specified.

3. Simulation results

This section describes the simulation model and presents the results obtained using the ns-2 network simulator [5]. In all simulations the data rate at physical layer is 11 Mbps and the routing protocol is DSR (Dynamic Source Routing [6]). We calculated a 90% confidence interval for mean. The intervals are represented by vertical bars.

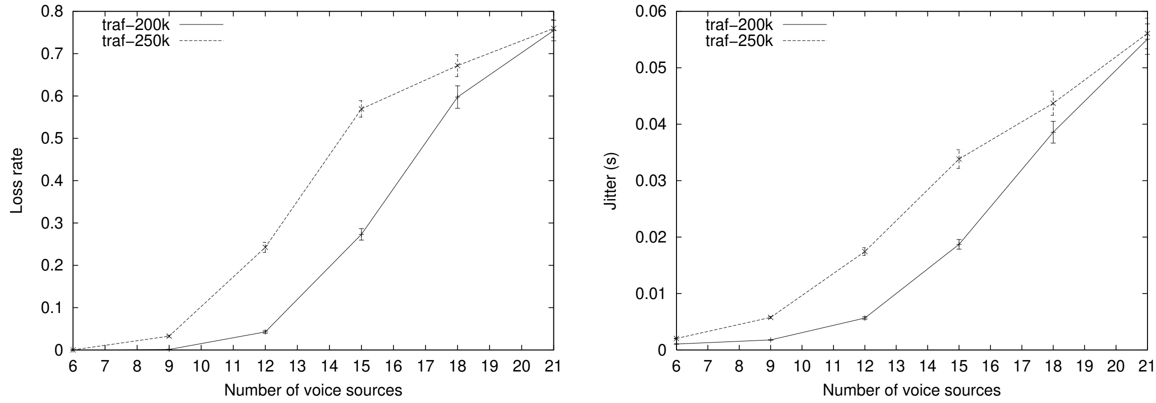
A two-state-Markov (On-Off) model is used to simulate voice sources with talk-spurts. On and Off states are modeled by random variables exponentially distributed with mean values 1.2 s and 1.8 s, respectively ([7, 8]). During On periods voice traffic is modeled by a CBR source at 64 kbps, with packets of 160 bytes, simulating Pulse Code Modulation (PCM) voice [1]. A background traffic is modeled by five CBR sources sending packets of 500 bytes at 200 kbps and 250 kbps, simulating low and medium load conditions, respectively. The simulation time is 400 s and the starting time of each source is uniformly distributed between 1 s and 11 s.

All packets have 250 ms of lifetime, beyond which a packet is considered lost. Thus, the packet that takes more than 250 ms to arrive is discarded by the recipient. We do not take into account the coding/decoding delay, the packet generation delay, and the queuing delay at the recipient. For PCM encoding, delivery rate should never drop under a percentage of 95% of all generated packets, to prevent significant loss in quality [2].

3.1. Single hop networks

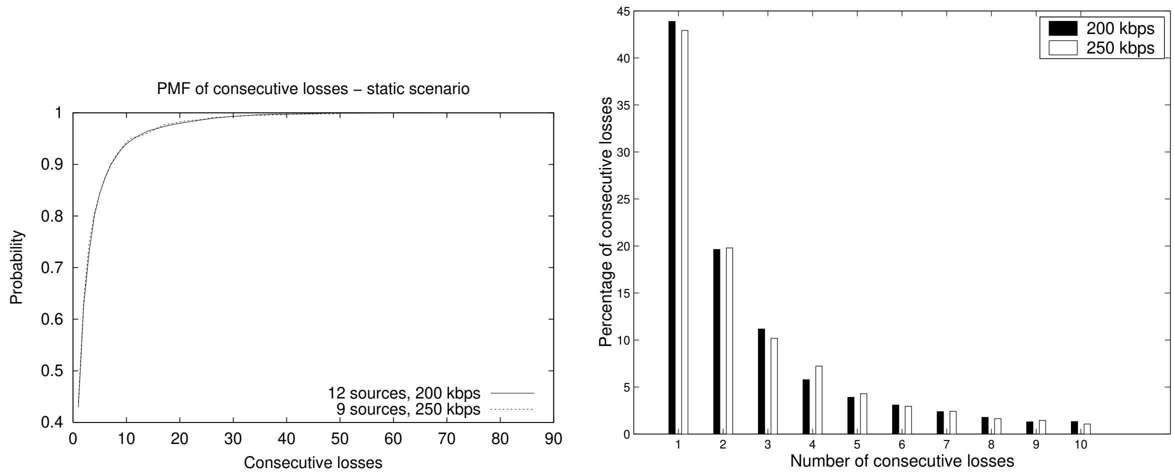
This subsection presents results related to the effect of mobility on voice traffic in single hop ad hoc networks. In this kind of network there is no need for routing because all nodes can communicate directly with each other. Thus, an increase of network load affects exclusively the medium access time, which might lead to an increase of packet loss. The simulation scenario is composed of 40 fixed nodes with transmission range of 250 m in a 150 m $\times$ 150 m area.

Figure 1(a) presents the impact of increasing background traffic on loss rate. The results show that under a low load condition we can have 12 voice sources transmitting at the same time, while under medium load only 9 voice sources can transmit. This represents a capacity degradation of 25% when the network load is increased by 25%. For 12 voice sources under a low load and 9 voice sources under a medium load the jitter remained almost the same, as shown by Figure 1(b).



(a) Loss rate (b) Jitter

Figure 1: Effect of network load on voice transmission.



(a) PMF of consecutive losses (b) Distribution of consecutive losses

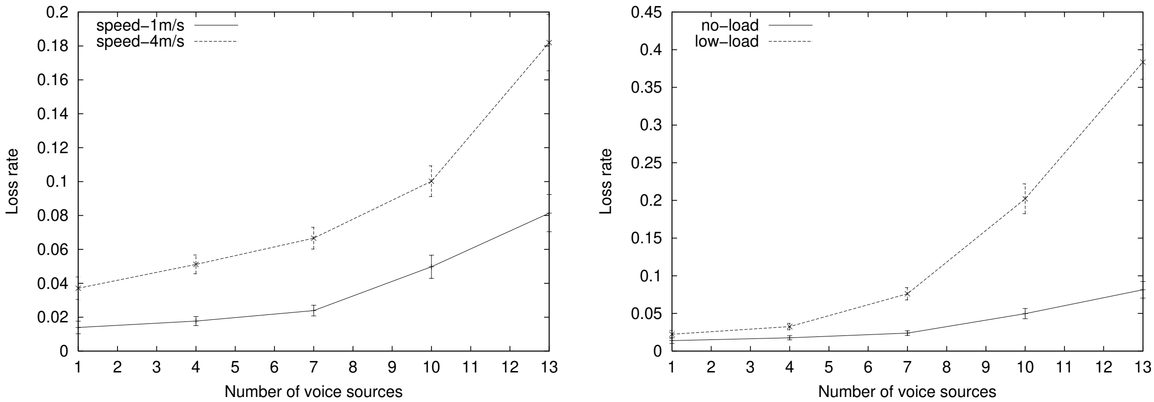
Figure 2: Effect of network load on consecutive losses.

Figure 2(a) illustrates the PMF (Probability Mass Function) of the consecutive packet losses for 12 voice sources under a low load and 9 voice sources under a medium load. In Figure 2(a), the two curves behave almost the same. It occurs due to the fact that packet losses are mainly affected by the medium access mechanism. Figure 2(b) shows the distribution of burst losses up to ten consecutive losses. The curves for 200 kbps and 250 kbps are very similar. In addition, one single loss and two consecutive losses represent most of losses and 80% of all burst losses are less or equal to 4 consecutive losses.

3.2. Multihop networks

In this subsection we present results related to the effect of mobility and node density on the capacity of voice transmission in multihop ad hoc networks. The scenario consists of 40 nodes with transmission range of 250 m in a 800 m $\times$ 600 m area, which provide a well connected scenario with a $1/12000$ m^2 node density. It was defined two mobility levels: low and medium, with average speed (vm) of 1 m/s and 4 m/s, respectively. Node speed is uniformly distributed in the following interval: $0.8vm \leq v \leq 1.2vm$. We simulated zero and low load conditions for both mobility levels. In these specific simulations, background traffic was modeled by 20 CBR sources at 16 kbps.

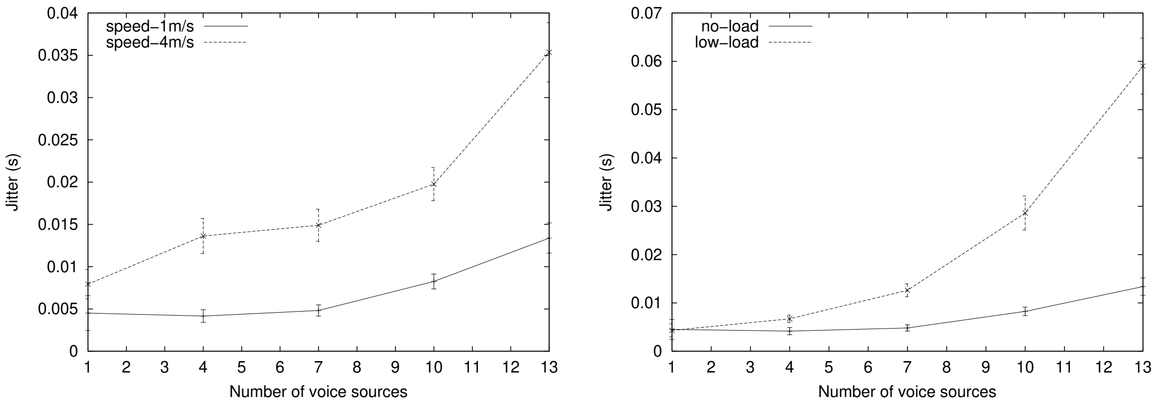
Figures 3(a) and 3(b) show the influence of mobility on network capacity according to the number of voice sources. In a zero load condition we can have 10 voice sources transmitting simultaneously for a low mobility and 4 voice sources for a medium mobility, while in a low loaded network with low mobility we can have 5 voice sources. It shows that mobility has a great impact on network capacity.



(a) No load

(b) Speed = 1 m/s

Figure 3: Effect of mobility on loss rate.



(a) No load

(b) Speed = 1 m/s

Figure 4: Effect of mobility on jitter.

Figures 4(a) and 4(b) illustrate the influence of mobility on jitter, emphasizing the capacity degradation due to the increase of load and mobility. An interesting observation is that jitter has increased more than 40% considering 10 voice sources at a low mobility and 4 voice sources at a medium mobility (Figure 4(a)). When we fixed the speed and varied the background traffic jitter remains almost constant, considering 10 voice sources transmitting under zero load and 5 voice sources under low load. This result is very similar to the one obtained for single hop networks when backgroung traffic was increased.

Figure 5 presents the PMF of consecutive losses for a scenario with no load and average speed of 1 m/s. The capacity of this scenario is 10 voice sources (Figure 3(a)) and the loss rate for 7 sources and 13 sources is 2.4% and 8.1%, respectively. Figure 5 shows that as the number of voice sources increases the PMF grows faster to one. Although the number of consecutive losses is represented up to 250, it can achieve almost 2,000 while in a single hop network this number is less than 90 consecutive losses.

Figure 6(a) indicates that for a low loss rate (7 sources) the probability of occurance of one single packet

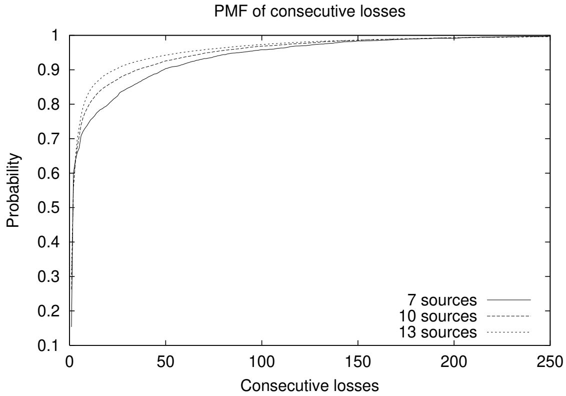
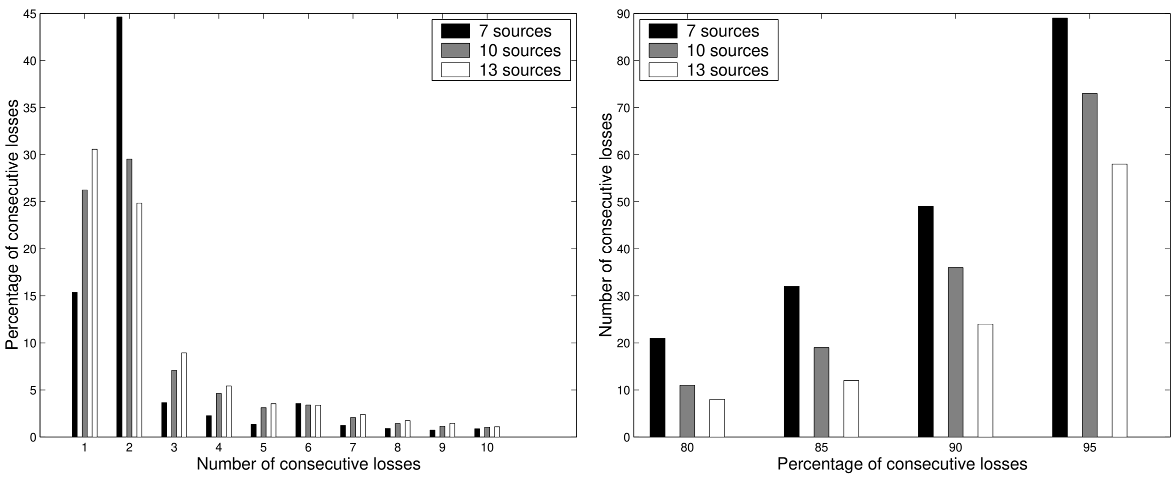


Figure 5: Consecutive losses 1 m/s and no load.

loss is smaller, because most of the packet losses occurs due to changes on topology. It also shows that as the number of voice sources increases the distribution gets closer to the expected one. Figure 6(b) makes the difference even more clear. It shows that for 10 voice sources 80% of the number of burst losses are less than 12 consecutive losses and only 5% are more than 73 consecutive losses.



(a) Distribution of consecutive losses

(b) Percentage of consecutive losses

Figure 6: Consecutive losses - 1 m/s and no load.

Figure 7 presents the PMF of the consecutive losses from scenarios with low mobility and medium mobility. It can be noticed that the curve with low mobility grows faster than the other, proving that topology changes has a great impact on consecutive losses, but the difference between the scenario with low mobility and medium mobility is small. There is a considerable difference between no load and low load because of the loss rate. For 4 voice sources under a low load the loss rate is less than 3% (Figure 3(b) while the other two curves correspond to a 5% loss rate.

Figure 8(a) and 8(b) compare node mobility in a scenario without background traffic. From Figure 8(a) we can see that a single packet loss and two consecutive losses correspond to more than 56% of the total and most of the losses occurs up to 10 consecutive losses. Figure 8(b) shows that 80% of the burst losses is less than 12 consecutive losses for low mobility and less than 17 for medium mobility.

We also address another important issue concerning the cause for packet losses. First, we separated lost packets in two groups according to the loss cause. The first group, named Lifetime, includes all packets dropped by the recipient due to lifetime expiration, previously defined as 250 ms. The other group, named Others, contains

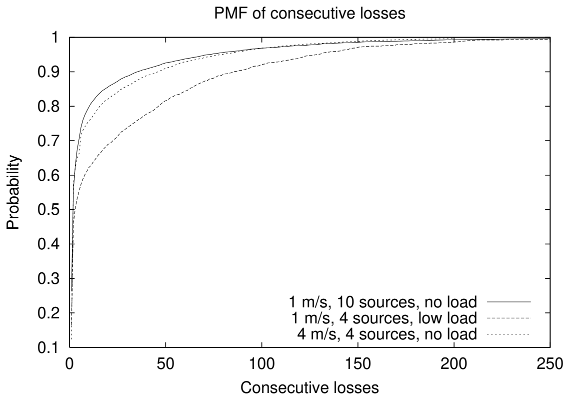
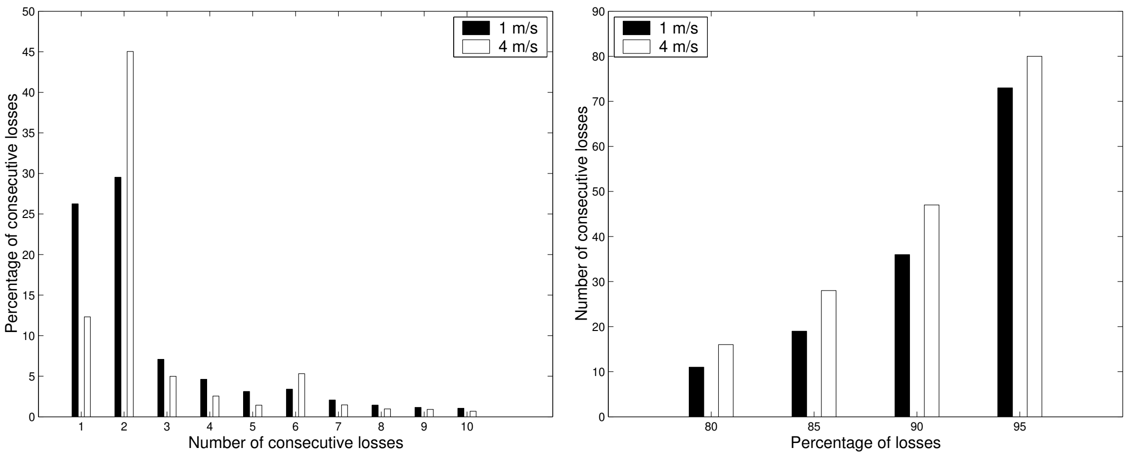


Figure 7: Effect of mobility on consecutive losses.



(a) Distribution of consecutive losses

(b) Percentage of consecutive losses

Figure 8: Effect of mobility on consecutive losses.

packets lost for any other reason, such as collision, no route, MAC queue overflow, etc.

Tables 2 and 3 summarize the influence of mobility and network load on the percentage of losses separated on groups. The increase of network load implies larger medium access time, which causes an increase in the number of packet losses due to lifetime expiration, despite the Lifetime percentage decrease. On the other hand, as mobility increases the percentage of packet loss due to other reasons is larger, indicating that mobility has greater impact on the second group than network load. This is expected because mobility reduces the packet delivery rate of routing protocols [9].

Table 2: Loss cause with no load.

	Loss cause			
Speed	Lifetime (%)	total	Others (%)	total
1 m/s	69.14	3315.3	30.86	1270
4 m/s	45.64	3095.4	54.36	3638.5

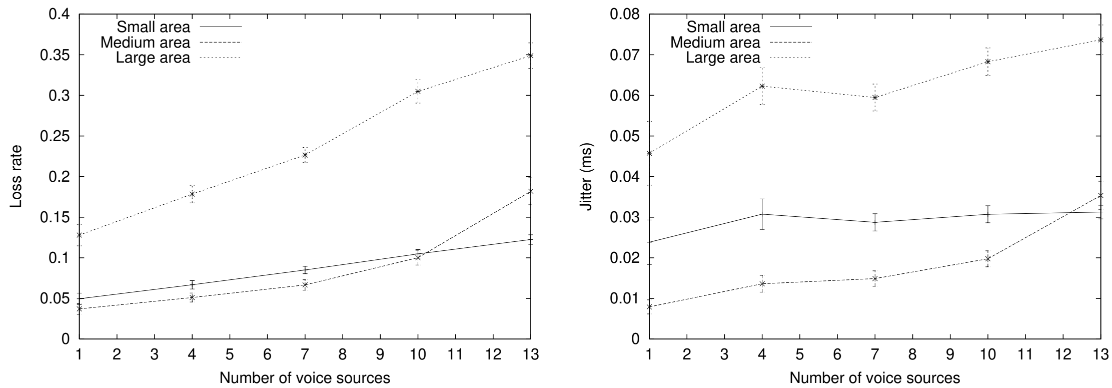
The node density affects the network connectivity, which is strongly related to the network capacity of voice transmission. Therefore, the following results refer to the effect of node density in multihop ad hoc networks. In order to analyze this effect, we generate two others scenarios in which the node density was changed by keeping the number of nodes constant and varying the simulation area. Thus, three different areas were simulated. The first one is a 600 m $\times$ 600 m field (small area), which represents a density of 1 node by 9,000 m^2. The second one is a 800 m $\times$ 600 m field (medium area), which represents a density of 1 node by 12,000 m^2. The last one is a 1,200 m

Table 3: Loss cause with speed 1 m/s.

	Loss cause			
Load	Lifetime (%)	total	Others (%)	total
zero	69.14	3315.3	30.86	1270
low	68.43	14.434.3	31.57	8614.1

$\times$ 500 m field (large area) which represents a density of 1 node by 9,000 m^2. The average speed is set to 4 m/s.

Both loss rate and jitter appear to have a similar behavior (Figure 9(a) and 9(b)). Due to the higher probability of link failures and the lack of connectivity, the large area cannot support voice traffic according to the QoS parameter of 95% for packet delivery. The medium area performs better than the small area with a small number of voice sources, because of the effect of the medium access contention in the small area.



(a) Loss rate

(b) Jitter

Figure 9: Effect of node density on voice transmission.

Figure 10 shows that as the area decreases the PMF grows faster, which means that the network connectivity has direct influence on consecutive losses. Therefore, a network with a high level of connectivity and with a small maximum number of hops can reduce burst losses.

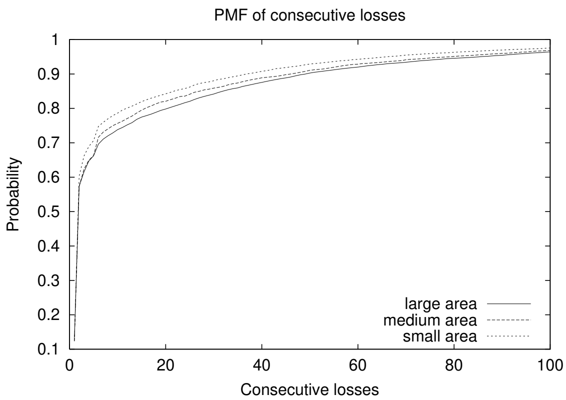
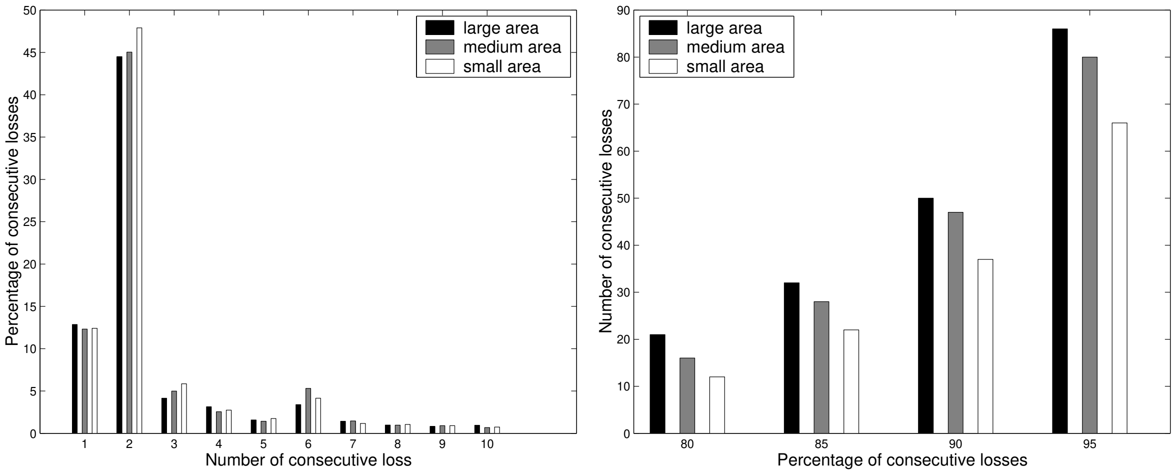


Figure 10: Effect of node density on consecutive losses - 4 voice sources.

Figure 11(a) reveals that one single packet loss and two consecutive losses correspond to most of the total burst losses. Figure 11(b) shows that 80% of the burst losses corresponds to less than 17 consecutive losses in the medium area and less than 13 in the small area, while 95% of burst losses corresponds to less than 81 consecutive losses in the medium area and less than 67 in the small area.



(a) Distribution of consecutive losses

(b) Percentage of consecutive losses

Figure 11: Percentage of consecutive losses - 4 voice sources.

4. Conclusions

This paper analyzed the effect of mobility and node density on voice transmission capacity in ad hoc networks, more precisely the influence of mobility on loss rate, jitter, and consecutive losses. The results show that the increase of mobility and network load degrade the network capacity in different ways. Network load directly affects the medium access time causing packet losses due to lifetime expiration, while mobility affects other parameters related to routing, which enlarge packet losses. The increase of mobility causes a large reduction in voice transmission capacity in multihop ad hoc networks. The results reveal that changes in the network topology have a great impact on burst losses, since node density is strongly related to the network connectivity. Thus, high connectivity and a small number of hops reduces burst losses.

References

[1] A. Köpsel, J.-P. Ebert and A. Wolisz, "A performance comparison of point and distributed coordination function of an IEEE 802.11 WLAN in the presence of real-time requirements", in *Workshop on Mobile Multimedia Communications (MoMuC2000)*, Tokyo, Japan, Oct. 2000.

[2] A. Köpsel and A. Wolisz, "Voice transmission in an IEEE 802.11 WLAN based access network", in *Workshop on Wireless Mobile Multimedia (WoWMoM'2001)*, Rome, Italy, July 2001.

[3] P. B. Velloso, M. G. Rubinstein and O. C. M. B. Duarte, "Analyzing voice transmission capacity on ad hoc networks", in *International Conference on Communications Technology (ICCT 2003)*, Beijing, China, 2003.

[4] ITU-T, "One-way transmission time", in *Recommendation G.114*, Mar. 1993.

[5] K. Fall and K. Varadhan, *The ns Manual*. UC Berkeley, LBL, USC/ISI, and Xerox PARC, Apr. 2002. Available at http://www.isi.edu/nsnam/ns/ns-documentation.html.

[6] D. B. Johnson, D. A. Maltz, Y.-C. Hu and J. G. Jetcheva, *The Dynamic Source Routing Protocol for Mobile Ad Hoc Networks (DSR)*, Feb. 2002. Internet draft.

[7] P. Brandy, "A technique for investigating on/off patterns of speech", *Bell Labs Tech Journal*, vol. 44, no. 1, pp. 1–22, Jan. 1965.

[8] A. Markopoulou, F. Tobagi and M. Karam, "Assessment of VoIP quality over internet backbones", in *Proceedings of the IEEE Conference on Computer Communications (INFOCOM)*, New York, USA, June 2002.

[9] S. R. Das, C. E. Perkins and E. M. Royer, "Performance comparison of two on-demand routing protocols for ad hoc networks", in *Proceedings of the IEEE Conference on Computer Communications (INFOCOM)*, pp. 3–12, Mar. 2000.