Analyzing voice transmission capacity on ad hoc networks

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***Abstract*— This paper analyzes voice transmission capacity on ad hoc networks by performing simulations related to delay and jitter. We evaluate the influence of QoS provision and mobility on the number of voice transmitting sources. Results show that the maximum number of voice transmissions can be increased when medium access time is reduced by means of a service differentiation mechanism applied to the MAC layer. Also, mobility and network load variations degrade the network capacity for voice transmission, mainly on multihop mobile networks.**

I. Introduction

Nowadays, wireless communication plays an important role in computer networks, due to its low implementation cost and high flexibility. Thus, wireless local area networks (WLAN) are becoming common place but transmitting real-time traffic in such networks 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 they can be an ad hoc network, which is characterized by no infrastructure, where each node communicates 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, they are most convenient in places where there is no infrastructure and it is too expensive to build it or in places where 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 (MAC) mechanisms and deal with exposed and hidden terminal problems, adding considerable complexity to nodes, especially in multihop networks, where they also act as routers. Besides, ad hoc networks must cope with other wireless medium 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.

The transport of real-time traffic must fulfill some QoS requirements which are specific to real-time applications. Our goal is to analyze the capacity of voice transmission in ad hoc networks considering parameters such as delay and jitter. There are some works related to voice transmission in IEEE 802.11 networks but only in the infrastructured mode. Köpsel *et al.* [1] analyzed DCF and PCF 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. They showed that PCF performs better in high loaded networks and that increasing BER degrades network capacity.

The research on QoS support in ad hoc networks includes QoS models, resource reservation signaling, QoS routing, and QoS on MAC sublayer. In [3], the main aspects related to QoS on MAC and the performance of three service differentiation schemes for IEEE 802.11 with TCP and UDP flows are presented.

The main contribution of our work is the analysis of the capacity of voice transmission in ad hoc networks and the evaluation of QoS provision impact on voice traffic. The remainder of this paper is organized as follows. Section II briefly summarizes the operation of 802.11 and presents the main technologies for providing QoS on MAC 802.11 networks. Simulation details and results are shown in Section III. Section IV presents our conclusions.

II. QoS in ad hoc networks

Real-time voice traffic have QoS requirements such as bounded end-to-end delay, maximum jitter, and limited loss rate. Delay consists of four basic components and plays an important role in loss of interactivity. It includes code/decode delay, packet generation delay, propagation delay, and queuing delay. Table I presents some tolerance to delay reference values recommended by the ITU-T [4].

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 sink with the same

TABLE I

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

temporal relation 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 sink.

The IEEE 802.11 standard [5] includes physical (PHY) and link layer specifications. At the link layer, two MAC methods are available. There is one basic mechanism (Distributed Coordination Function - DCF), which supports infrastructureless networks and a centralized mechanism (Point Coordination Function - PCF), which can be considered an extension to DCF, specified to support real-time traffic.

DCF is a distributed mechanism based on CSMA/CA in which every station must sense the medium before transmitting any frame. If the medium is idle the station must wait for DIFS (Distributed Inter-Frame Space) units of time. Then, the sender should wait for a random time interval (backoff) between zero and the maximum contention window (backoff = [0,max_CW]). By the end of the backoff time, the sender can finally transmit. Backoff is part of the collision avoidance mechanism. Due to significant signal attenuation, wireless nodes are not capable of detecting collision at the recipient, but only at the sender. Thus, the sender must wait for an ACK frame. In order to provide priority to ACK over data frames, the recipient has to wait for SIFS (Short Inter-Frame Space), an amount of time smaller than DIFS, before sending the ACK.

There are three main schemes for providing service differentiation in ad hoc networks based on IEEE 802.11, which consist of assigning different values to specific parameters of the DCF mechanism [3]. The first one changes the backoff function in such a way that nodes with higher priority have a smaller maximum contention window value. Another technique, in a similar way, assigns different values of DIFS to each node according to its required priority. In that case, the station with higher priority has a smaller DIFS value. The last one consists of assigning a larger maximum frame length to nodes with higher priority. The first two techniques achieve service differentiation by reducing the medium access time, while the last one increases the amount of data transmitted in each frame.

In this paper we consider the first mechanism in order to analyze through simulations the influence of QoS provision on voice transmission capacity in ad hoc networks.

III. SIMULATION RESULTS

This section describes the simulation model and presents the results obtained using the ns-2 network simulator [6]. In all simulations the data rate at PHY layer is 11 Mbps and the routing protocol is DSR [7].

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 ([8], [9]). 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 400s 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. For PCM encoding, delivery rate should never drop under a percentage of 95% of all generated packets, to prevent significant loss in quality [2].

A. QoS provision

This subsection presents results related to QoS provision in ad hoc networks based on IEEE 802.11. We defined three priority levels: no priority, low priority, and high priority, assigning larger maximum contention window values to background traffic sources. The assigned values to background traffic are CW, 2CW, and 3CW, respectively. This means that voice sources have a constant maximum contention window value (CW), while the maximum contention window for background traffic sources varies from 1 to 3 times CW, according to the priority level.

In order to assess the effect of this QoS technique, we chose a simple scenario in which the routing effect is minimized. This scenario is composed of 40 fixed nodes with transmission range of 250 m in a 150 m $\times$ 150 m area, which means that packets do not need to be routed because nodes can communicate directly with each other.

Figures 1 and 2 show that varying the maximum contention window allows the increase of voice transmission capacity. Under a low load condition it was possible to augment by two, the number of voice sources with low priority and by four with high priority. It can also be noticed that a better differentiation is obtained under a medium load condition.

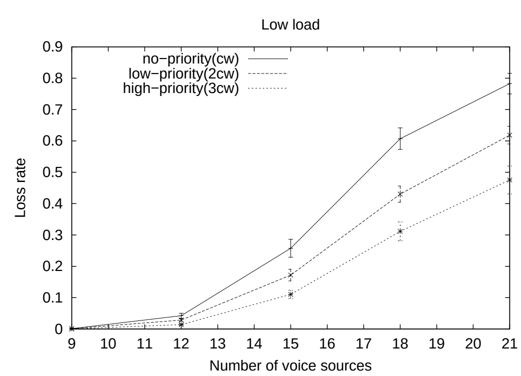

Fig. 1. Effect of QoS on loss rate under low load.

Figures 3 and 4 show the jitter behavior when varying maximum contention window value. In this case, the jitter

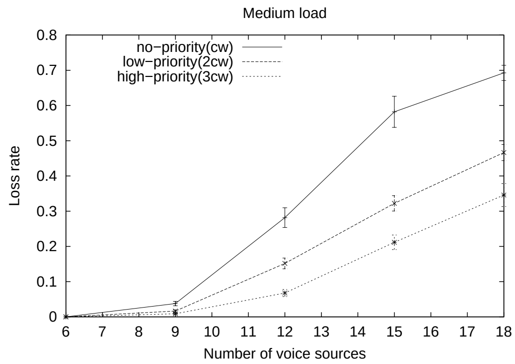

Fig. 2. Effect of QoS on loss rate under medium load.

appears to be less sensitive to QoS provision than the loss rate. These results from QoS provision point out that there is no clear relation between the maximum contention window size, the network load, and the level of service differentiation obtained.

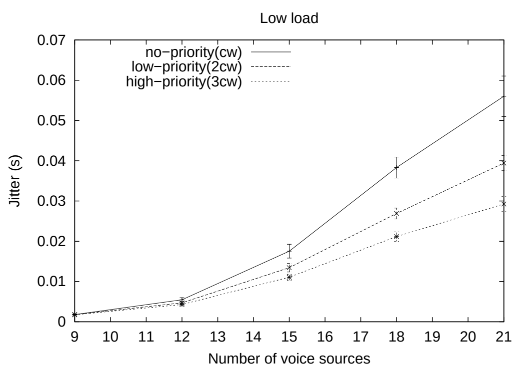

Fig. 3. Effect of QoS on jitter under low load.

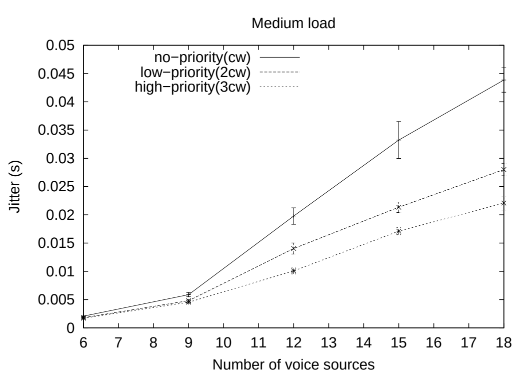

Fig. 4. Effect of QoS on jitter under medium load.

B. Mobility

In this subsection we present results related to the effect of mobility on voice transmission capacity in 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. We chose 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: $0.8vm <= v <= 1.2vm$. 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 5 and 6 show the influence of mobility on network capacity according to the number of voice sources. In a zero load condition we can have eight voice sources transmitting simultaneously for low mobility and two for medium mobility, while in a low loaded network with low mobility we can have four voice sources. It shows that mobility has a great impact on network capacity.

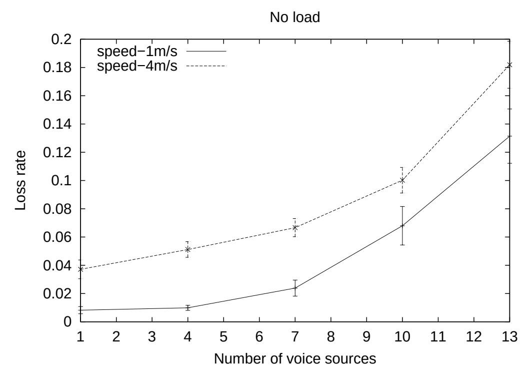

Fig. 5. Effect of mobility on loss rate without load.

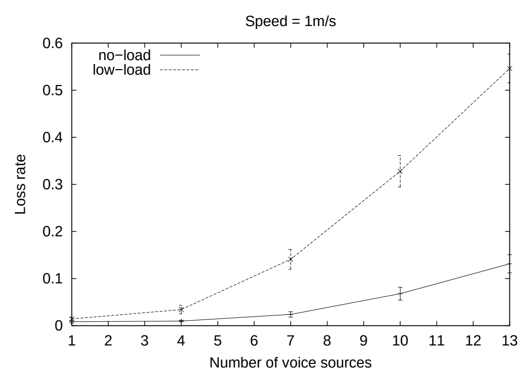

Fig. 6. Effect of mobility on loss rate with speed = 1 m/s.

Figures 7 and 8 present the influence of mobility on jitter, emphasizing the capacity degradation due to the increase of load and mobility. An interesting observation is that jitter is more sensitive to load variations than loss rate, considering that jitter had a larger variation as load increased. On the other hand, loss rate is more sensitive to mobility variations than jitter.

We also address another important issue concerning the cause for packet losses. First, we separated lost packets in

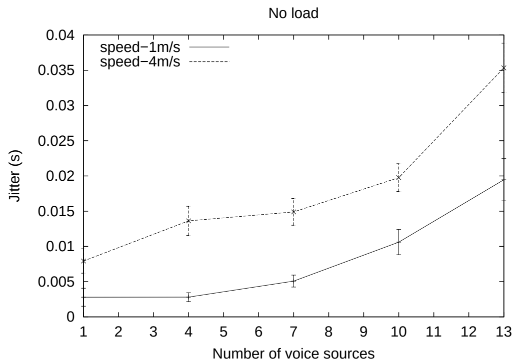

Fig. 7. Effect of mobility on jitter without load.

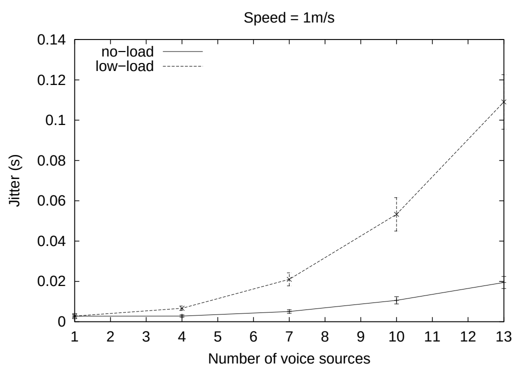

Fig. 8. Effect of mobility on jitter with speed = 1 m/s.

two groups according to the loss cause. The first group, named Lifetime, includes all packets lost due to lifetime expiration, previously defined as 250 ms. The other group, named Others, contains packets lost for any other reason, such as collision, no route, MAC queue overflow, etc.

Tables II and III 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 [10].

TABLE II

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

IV. CONCLUSIONS

Wireless networks have high bit error rate, frequent changes in link conditions, and restrictions related to bandwidth and energy consumption. Besides all these constraints ad hoc networks have no infrastructure to support mobility and QoS, which implies the increase of node complexity, making difficult the transmission of real-time traffic.

TABLE III

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

This paper analyzed voice transmission capacity in ad hoc networks, more precisely the influence of mobility and QoS provision. The service differentiation technique evaluated in this paper consists of assigning different maximum contention window values to each kind of source.

The results show that the increase of mobility and network load degrade 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. Jitter is more sensitive to variations in network conditions than loss rate, except for mobility. Nevertheless QoS provision had more influence on loss rate than on jitter. The increase of network load causes a large reduction in voice transmission capacity in multihop ad hoc networks.

In spite of the capacity improvement for voice transmission achieved by using service differentiation, it is important to develop a distributed mechanism of connection admission control to avoid voice traffic capacity degradation.

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