An Adaptive MAC Scheme to Support VoIP Across Ad Hoc IEEE 802.11 WLANs

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Abstract: This article describes the performance enhancements of the Ad hoc IEEE 802.11 WLANs (Wireless LANs) for VoIP support. The presented strategy is a result of the new outlook on the IEEE 802.11 networks capabilities and performance enhancement. The proposed solution is dedicated to real time services support and is based on the concept of the adaptive MAC scheme. A method of the available bandwidth measurement and estimation was introduced. Novel procedures were developed and applied in order to shape the real time traffic and to provide the Call Admission Control (CAC) mechanism. Large scale simulations for different numbers of Voice over IP (VoIP) sources and various voice codecs have been carried out. They demonstrate the efficiency of CAC mechanism and the increase of the network capacity for VoIP.

Keywords: Ad-hoc networks, bandwidth measurement, Voice over Wi-Fi

1. Introduction

Wireless networks operating in ad hoc mode can be a very promising solution for public-safety or search-and-rescue operations. The possible scenario is to use the ad hoc networks in a state of emergency, e.g., natural disasters or terrorist attacks, when the communications infrastructure is damaged. An important issue for such a network is the ability to support cooperation between two or more emergency services, e.g., the fire brigade, police squad, rescue team, medical service [1,2]. On the other hand, the performance of the network decreases as the number of wireless users grows. For that reason, a smart mechanism should be introduced, which allows topology control and network scalability [3]. When considering the hierarchical structure of the command system of the emergency services, different ranks of users should be taken into account. This affects the priority of users, as well as the type of allowed services.

Among many wireless solutions, IEEE 802.11 networks appear to be the most popular. Although the most common weakness of WLANs is the insufficient support
of the real time services [4,5], the VoWiFi (Voice over WiFi) is gaining increasing attention in literature. The influence of the MAC protocol parameters, the coding rate and packetization interval [6-8,13] on the network performance were analyzed and possible enhancements were proposed in a number of papers.

Although there are many methods for improving network efficiency [9-12], the question as to how to guarantee the quality of services and how to provide an efficient CAC mechanism still remains open. The present article is an attempt to fill this gap.

2. Idea and assumptions

In the case under consideration, the aim of the network optimization is to increase the number of VoIP streams. The assumed network operates in ad hoc mode and consists of small group of users, e.g., the fire brigade or rescue team. They typically employ voice communication in emergency situations. Therefore, the authors made an assumption that only VoIP service is used. Users have different ranks, which determines some differences between priorities. Thus, the trade off between the available bandwidth, the allowed number and the rank of users has been introduced intentionally.

The network model assumes a WLAN based solution. The authors decided to use the IEEE 802.11b standard as it provides good throughput and modulations more resistant to interferences, which is a real advantage of the network operating in ad hoc mode. The MAC QoS mechanisms defined in the IEEE 802.11e standard [6] were also taken into account. These mechanisms are a good starting point to enable prioritization and bandwidth reservation [10-12]. In particular, the authors introduced the adaptation of a Contention Window (CW) size to the type of frame and the rank of the user.

![Figure 1. The idea of VoWiFi optimization](image-url)
Figure 1 illustrates the concept of WLAN optimization for VoIP support. The available bandwidth level is the main factor allowing assessment of the traffic load in the network. Cross-layer mechanisms are crucial for network performance improvement. They enable the RT traffic shaping or MAC adaptation where the available bandwidth is too small or if the level of service is not satisfactory. The CAC mechanism prevents new VoIP calls where the available bandwidth level is too low.

In the proposed solution, the network consists of different rank users. For the sake of simplicity, high rank users shall be denoted as special users while the rest shall be referred to as regular users. This perfectly corresponds to the scenario of humanitarian aid, when volunteers assist the various services. Another example is a situation when the fire brigade, police and civilians cooperate within small groups while strengthening an embankment during a flood. However, if the available bandwidth is too small, special users prevail over the network. Eventually, the lowest rank users can be completely blocked.

A separate question is the assessment of the network resources, e.g. channel utilization. Since ad-hoc networks are bandwidth limited, not all measurement methods can be applied. Therefore we developed and implemented a method proposed by Sarr et al. [14], which is based on permanent measurement of channel utilization [15].

Figure 2 illustrates the extended protocol stack with cross-layer interactions. Bandwidth Prediction Control Protocol (BPCP) allows the monitoring of parameters in the physical layer, to measure the channel utilization level and also to switch MAC protocol states, as explained in the subsequent section. RT traffic shaping relies on codec negotiation and audio packets aggregation. Closed Network Mode is based on the concept of the Resource Manager that controls network traffic [15,16].
3. MAC protocol states

At the beginning of the operation, special stations can cooperate with regular stations and use standard access schemes, until BPCP detects the insufficient bandwidth and initiates the Acquisitive Mode (AM), Figure 3.

![Figure 3. MAC protocol states](image)

During AM mode, the Backoff interval is minimized according to the rank of the user. As a result, special stations prevail over the network. Only a small part of the bandwidth can be hard-won by remaining users. To determine the Backoff interval, the Contention Window parameter is used, however different values have been introduced, depending on the rank of the user and the type of frame (Control, Data, Broadcast or RTData). During the AM mode BPCP forces RT traffic shaping and also plays a CAC role, i.e., prevents new calls where a too small bandwidth is detected. The AM mode is initiated independently and locally by the station which detected insufficient bandwidth level.

If the available bandwidth is still too small, BPCP triggers a mechanism called the Network Self-Organizing Mechanism, which is responsible for creating a Closed Network Mode [15].

4. IEEE 802.11 WLAN performance enhancement

The performance of the IEEE 802.11 network depends strongly on the audio packet size and arrival frequency, as well as on the total overhead introduced by protocol headers and PHY layer [4-8]. Table 1 presents the maximum number of VoIP calls for standard MAC parameters, different data rates and various audio codecs [7].

With a view of increasing network performance, a set of optimization techniques has been proposed. One of them relies on real time traffic shaping. This can
be done by negotiating of the voice codec, e.g., switching to the codec with a lower coding rate if the available bandwidth is too small. Another way is to aggregate audio packets in the application layer. As a result, audio packets are generated not as frequently, while the payload is increased, i.e., doubled or tripled.

<table>
<thead>
<tr>
<th>Voice codec</th>
<th>Data rate [Mbit/s]</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1</td>
</tr>
<tr>
<td>G.711-64</td>
<td>7.4</td>
</tr>
<tr>
<td>G.726-32</td>
<td>9.7</td>
</tr>
<tr>
<td>G.728 A-16</td>
<td>15.8</td>
</tr>
<tr>
<td>GSM 6.10</td>
<td>11.9</td>
</tr>
<tr>
<td>G.729</td>
<td>12.7</td>
</tr>
<tr>
<td>G.723.1A</td>
<td>18.8</td>
</tr>
</tbody>
</table>

The CW size adaptation was introduced for packets prioritization in accordance with the type of service. The CW is shortened if the service has higher priority. This mechanism also allows the prioritization of users, if only one type of service is assumed, e.g., VoIP.

5. Simulation scenarios

In order to assess the influence of the proposed optimization mechanisms on network performance, a series of simulations were performed with the employment of the OMNET++ v4.0 simulation tool. In particular, we investigated the advantage of employing the following mechanisms: the codec negotiation, audio packets aggregation, as well as the CW size adaptation.

The following parameters were assumed: commercial voice codec; typical protocol headers (MAC header = 30B, IPv4 header = 20B, UDP header = 8B); free space propagation model and lack of mobility. The data rate was set at 2Mbit/s. Values of the MAC protocol parameters are shown in Table 2.

<table>
<thead>
<tr>
<th>DIFS</th>
<th>SIFS</th>
<th>CW(_{\text{min}})</th>
<th>CW(_{\text{max}})</th>
<th>Data rate</th>
<th>PHY header</th>
<th>MAC header</th>
<th>ACK</th>
</tr>
</thead>
<tbody>
<tr>
<td>50μs</td>
<td>10μs</td>
<td>31</td>
<td>1023</td>
<td>2Mbit/s</td>
<td>192μs</td>
<td>34 bytes</td>
<td>304μs</td>
</tr>
</tbody>
</table>

Commercial G.711 voice codec was chosen, as it is the most frequently used in the literature of the topic. With a view of evaluating the codec negotiation mechanism, additional codecs were selected, namely G.726 and G.728. Their attributes are shown in Table 3.
Table 3. Codecs attributes

<table>
<thead>
<tr>
<th>Codec</th>
<th>G.711</th>
<th>G.726</th>
<th>G.728</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bit rate [kbit/s]</td>
<td>64</td>
<td>32</td>
<td>16</td>
</tr>
<tr>
<td>Framing interval [ms]</td>
<td>20</td>
<td>20</td>
<td>30</td>
</tr>
<tr>
<td>Payload [B]</td>
<td>160</td>
<td>80</td>
<td>60</td>
</tr>
<tr>
<td>Packets/sec</td>
<td>50</td>
<td>50</td>
<td>33,3</td>
</tr>
</tbody>
</table>

6. Simulation results

Audio codec negotiation

In this scenario we evaluated the influence of the audio packets coding rate on network performance. The results of simulation are given below.

![Graph showing the average delay vs. the number of VoIP streams for various voice codecs: G.711 (64kbit/s), G.726 (32kbit/s) and G.728 (16kbit/s)]

Figure 4. The average delay vs. the number of VoIP streams for various voice codecs: G.711 (64kbit/s), G.726 (32kbit/s) and G.728 (16kbit/s)

![Graph showing the PLR vs. the number of VoIP streams for various voice codecs: G.711 (64kbit/s), G.726 (32kbit/s) and G.728 (16kbit/s)]

Figure 5. The PLR vs. the number of VoIP streams for various voice codecs: G.711 (64kbit/s), G.726 (32kbit/s) and G.728 (16kbit/s)
The average delay and the PLR vs. the number of VoIP streams are illustrated respectively in Figure 4 and 5. If G.711 codec is employed, 11 simultaneous VoIP calls can be established with an expected level of service quality. For more streams, the PLR and average packet delay grows rapidly. If G.726 codec is used, the maximum number of VoIP sources increases up to 14. This number reaches about 17 for G.728 codec.

**Audio packets aggregation**

In the second scenario, we investigated the advantage of applying the audio packet aggregation mechanism for G.711 codec for the following cases:
- Aggregation x1 (no aggregation);
- Aggregation x2 (two audio packets are combined);
- Aggregation x3 (three audio packets are joined together).

![Figure 6. The average delay vs. the number of VoIP streams for different audio packets aggregation levels](image)

![Figure 7. The PLR vs. the number of VoIP streams for different audio packets aggregation levels](image)

The average delay and the PLR vs. the number of VoIP streams are shown respectively in Figures 6 and 7. For the first case, no aggregation, the network capac-
ity reaches 11 VoIP streams. This number grows to 16 if two packets are combined and further increases up to 18 when three packets are aggregated.

**CW size adaptation**

In this scenario, we examined the advantage of introducing the CW adaptation mechanism. For the purposes of simulation, we used two types of stations, namely high priority stations denoted by A and regular stations denoted by B. The following parameters were assigned:

- station type A: $CW_{\text{min}} = 31$ and $CW_{\text{max}} = 1023$;
- station type B: $CW_{\text{min}} = 7$ and $CW_{\text{max}} = 15$.

We set the ratio between them as 50/50 which means, that 50% of them are type A and 50% are type B. Simulation results for G.711 voice codec are provided below.

![Graph showing average delay vs. number of VoIP streams](image1)

**Figure 8.** The average delay vs. the number of VoIP streams for stations A and B

![Graph showing PLR vs. number of VoIP streams](image2)

**Figure 9.** The PLR vs. the number of VoIP streams for stations A and B
The results presented in Figures 8 and 9 illustrate the advantage of the CW adaptation for user prioritization. If less than 10 VoIP streams are set, all stations can cooperate within the network and the level of average delay of packets and PLR remains acceptable. When new VoIP calls appear, regular stations observe higher packets delay and increased PLR. When the network consists of 16 nodes, nodes A detect further drops of service quality while nodes B can still be satisfactory.

**Triggering levels**

An important issue is the determination of the proper level of channel utilization for triggering between MAC states. The authors applied the Pareto optimization approach with a view of resolving this problem. Simulation results obtained for a 2Mbit/s data rate and G.711 voice codec can be found in [16].

Figure 10 shows the network throughput vs. traffic load. Triggering levels are also presented. If the network throughput reaches the limit denoted by B, the station switches from standard mode to AM. If it reaches another limit denoted by C, the station is not allowed to initiate a new call, as it inevitably will cause the decrease of service quality. For this reason, the station initiates Closed Network Mode and from this moment on, the network operates in point coordinated mode [15].

![Network throughput vs. traffic load](image)

**Figure 10. Network throughput vs. traffic load (triggering levels: A – switch from AM Mode to std., B – switch from std. to AM Mode, C – switch to Closed Network Mode)**
Figure 11 presents collisions vs. traffic load. The critical level of collisions is denoted by A. This information may be used additionally by BPCP to prevent new calls.

7. Conclusions

In this article, we presented the concept of the adaptive MAC scheme to support VoIP traffic in ad hoc WLANs for rescue operations. The issue of the network performance was discussed and optimization mechanisms were described. With a view of initiating subsequent states of the MAC protocol, a new method of the available bandwidth estimation was developed and triggering levels were assigned.

This new adaptive MAC scheme allows the initiation of optimization mechanisms or the prevention of new VoIP calls if the available bandwidth is too small. Eventually, it enables to switch the network to point coordinated mode.

For future research, it would be interesting to study the effect of the TCP traffic on the network capacity for VoIP. Based on this work, we will be investigating how to efficiently manage the network, where VoIP streams are combined with the TCP flows. Furthermore, we shall assess the efficiency of proposed extensions for the IEEE 802.11b network and compare it against the performance of the IEEE 802.11e WLANs.

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Chapter 6: Tactical Communications

REFERENCES


