A Resource Management Strategy Based on the Available Bandwidth Estimation to Support VoIP across Ad hoc IEEE 802.11 Networks

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Abstract — This paper focuses on performance evaluation to study the effects of the resource management strategy in ad hoc networks. 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 Resource Manager that organizes and controls the whole traffic in the network. Novel procedures were developed and applied in order to organize the network and manage the real-time traffic. A method for measuring and estimating available bandwidth was introduced. This method provides wireless station the capability of measuring the bandwidth independently of other stations in the network. Large-scale simulations for different numbers of voice sources and various voice codecs have been carried out. They show an increase of channel utilization reaching over 80% and significant growth of the network capacity, when synchronous transmission is applied.

Keywords - IEEE802.11 WLANs, ad-hoc networks, VoWiFi, resource management

I. INTRODUCTION

This paper deals with the issue of the resource management strategy in ad hoc wireless networks to support real-time services [1].

For over ten years a permanent development of IEEE 802.11 Wireless Local Area Networks (WLANs) is being observed [2]. Among the many advantages they offer, users appreciated the convenience and simplicity when accessing the network and establishing high data rate wireless connection. Thanks to the low-cost small-size devices, wireless networks seem to be ubiquitous. WLAN drivers are embedded in many different devices like notebooks, mobile phones, Personal Data Assistants (PDAs), cameras, etc. Despite the fact that WLANs were originally designed for data transport, today it is also demanded of them to be efficient for real-time services support.

Another advantage of WLANs results from the ad hoc mode, which is a method for wireless devices to directly communicate with each other. Operating in ad hoc mode allows all wireless devices within each other’s range to discover and communicate in a peer-to-peer manner without involving the central access point. This mode offers mobility and communications between users in areas without infrastructure or in all places with damaged infrastructure. From this point of view, WLANs operating in ad hoc mode can be a very promising solution for users, such as the fire brigade, rescue team, police squad or small military unit [3][4]. The possible scenario is to use the ad hoc network for public-safety or search-and-rescue operations.

An important issue for such network is the ability to support cooperation between two or more emergency services, e.g., the fire brigade, police squad, rescue team, medical service. On the other hand, it must be stressed that the performance of the network decreases as the number of wireless users grows. For this reason, a smart mechanism should be introduced, which allows topology control and network scalability [5]. The effect of the hidden node is one of the most difficult problems to solve, because it is intrinsic to the nature of the WLANs. The RTS/CTS mechanism is not recommended for the transmission of small packets, e.g., VoIP. A possible solution is to use an additional signaling channel, however it requires changes in the physical layer. This issue was widely discussed in [15][16].

When considering the hierarchical structure of the command system of the emergency services, different ranks of users should be taken into account. This will affect the priority of users, as well as the type of services allowed.

Although the most common weakness of WLANs is the insufficient support of the real-time services [6][7], the authors formulated a new outlook on the IEEE 802.11b network capability and possible performance enhancement. Despite the fact that there is a wide range of WLANs specifications, the issue of network optimization still remains open. QoS mechanisms were the subject of the IEEE 802.11e standard [8]. However, these mechanisms cannot guarantee the quality of services, although they slightly improve the network efficiency [9].

This paper presents the general concept of the resource management strategy and provides information on introduced procedures. All proposed mechanisms are integrated with each other and interact within an individual device as well as within the whole network.

The rest of the paper deals with the related work (Section 2), concept and assumptions (Section 3), the description of the proposed mechanisms (Section 4), simulation results and
their discussion (Section 5), conclusions (Section 6) and future work (Section 7).

II. RELATED WORK

The voice capacity of IEEE 802.11 networks is gaining increasing attention in the literature. Methods of voice traffic optimization, including voice codec negotiation, audio packets aggregation as well as the MAC protocol adaptation, can be found in many papers. In [9], the influence of the MAC protocol on the network performance was shown. This protocol operates in contention mode and thus inevitably introduces the PHY layer overheads, Backoff and protective periods, ACK frames and retransmissions in some cases. In [10], the authors analyzed the effect of the coding rate and packet size on the voice capacity of the Distributed Coordination Function (DCF).

In [11], dynamic Contention Window (CW) adaptation was suggested in order to minimize the number of collisions. The idea of the voice coding bit rate adaptation to the available network bandwidth was described in [23]. Experimental results confirmed the efficiency of the new scheme. The impact of different configuration parameters on the ad hoc network performance was presented in [24]. The following parameters were analyzed, the type of codec, packetization interval and the data rate. In [25], the authors presented the results of the capacity measurement of the IEEE 802.11e network for each access category. They also analyzed the effect of the TCP traffic on VoIP streams. In conclusion, they stated that 802.11e standard can protect the quality of VoIP if there is TCP traffic added. However, it cannot improve the capacity of the network.

Although proposed methods can improve network efficiency, the question as to how to guarantee the quality of services still remains open. In [14], the authors proposed to introduce additional signaling channel to inform other stations that the main channel is busy. This approach requires modification in physical and data link layers. Furthermore, there is still a lack of an efficient Call Admission Control (CAC) mechanism to protect voice traffic. In [13], a dynamic admission control mechanism was presented. This mechanism is based on the traffic analysis model in order to guarantee the QoS, however it is dedicated to the network with infrastructure.

III. CONCEPT AND ASSUMPTIONS

In the case under consideration, the aim of the network optimization is to get as high as possible number of VoIP streams with guaranteed voice quality. The assumed network operates in ad hoc mode and consists of small group of users, e.g., fire brigade or rescue team. In emergency situations they typically use voice communication. Therefore the authors made an assumption that there is only one type of service, namely VoIP.

Users of the network have different ranks, which enables to determine some differences between priorities. Thus, the tradeoff between the available bandwidth, the allowed number and the rank of users is introduced intentionally.

Figure 1. Performance enhancement of WLAN for VoIP support. [1]
and if they operate in promiscuous mode, they can receive and process all frames. The type of received frames (RTS, CTS, DATA, ACK) is recognized in the data link layer.

Knowing the bit rate and the length of received frames it is possible to calculate their transmission duration in the radio channel, denoted as $t_{AF}$ in (1).

$$t_{AF} = \frac{AF_{\text{length (bits)}}}{AF_{\text{bitRate (bits/sec)}}}$$

Having knowledge of $t_{AF}$ parameters, it is then possible to determine the channel utilization coefficient for the interval, e.g., from $t_i$ to $t_2$

$$U = \frac{t_{AF} + t_{rt} + \ldots + t_{AF} + n \cdot T_{ACK} + n \cdot SIFS + n \cdot DIFS}{t_2 - t_1}$$

where: $n$ is a number of frames; $t_{AF}$ denotes the duration of the $n$-th data frames; $T_{ACK}$ represents the ACK frame duration, Fig. 3.

When the current and the previous channel utilization levels are estimated, BPCP makes forecasts for the next period.

To validate the model of BPCP, enabling the measurement of the network throughput, theoretical analysis was performed and simulations were made using the OMNET++ v4.0 simulation tool with the INET Framework.

Fig. 4 shows the extended WLAN sublayer of the mobile node. This sublayer contains Throughput Meter In and Throughput Meter Out components to measure all incoming and outgoing traffic. This information is used to assess the total traffic load as well as the available bandwidth.

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**Figure 2.** BPCP alignment with protocol stack.

**Figure 3.** Transmission scheme in a contention mode of WLAN.

**Figure 4.** The extended WLAN sublayer of the mobile node.
TABLE I. MAC PARAMETERS

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>DIFS</td>
<td>50 µs</td>
</tr>
<tr>
<td>SIFS</td>
<td>10 µs</td>
</tr>
<tr>
<td>Slot Time</td>
<td>20 µs</td>
</tr>
<tr>
<td>CWmin</td>
<td>32</td>
</tr>
<tr>
<td>CWmax</td>
<td>1023</td>
</tr>
<tr>
<td>Data Rate</td>
<td>2Mbit/s</td>
</tr>
<tr>
<td>PHY header</td>
<td>192 µs</td>
</tr>
<tr>
<td>MAC header</td>
<td>34 bytes</td>
</tr>
<tr>
<td>ACK</td>
<td>304 µs</td>
</tr>
</tbody>
</table>

For the purposes of analysis and simulation, the following parameters were assumed: G.711 voice codec; typical protocol headers (MAC header = 30B, IPv4 header =20B, UDP header = 8B); free space propagation model and lack of mobility. The issue of mobility is crucial for NRM determination and is the topic of further study.

The values of the MAC and PHY parameters are listed in Table I. The main attributes of the G.711 codec are shown in Table II.

TABLE II. G.711 CODEC CHARACTERISTICS

<table>
<thead>
<tr>
<th>Codec</th>
<th>G.711</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bit rate [kbit/s]</td>
<td>64</td>
</tr>
<tr>
<td>Framing interval [ms]</td>
<td>20</td>
</tr>
<tr>
<td>Payload [B]</td>
<td>160</td>
</tr>
<tr>
<td>Packets/sec</td>
<td>50</td>
</tr>
</tbody>
</table>

The results of the simulation are presented in Fig. 5. Normal distribution of a throughput estimator was assumed, as well as a confidence interval with α=0.1 and β=1.64 (for cumulative distribution function equal to 0.9).

The period of time required for the transmission of one data frame and the acknowledging frame takes nearly 1.8ms. For that reason it is possible to send 11 acknowledged frames during one second. Audio packets are generated by codec periodically every 20ms. Assuming that stations work synchronously, i.e., after the first one had transmitted a packet, the second one generates it, then it is possible to obtain the network throughput equal to 1.3Mbit/s. Fig. 5. Higher traffic load will cause an increase of the collision rate and a drop in network efficiency.

![Graph](image)

Figure 5. Wi-Fi network throughput - contention mode, data rate 2Mbit/s.

B. The Network Throughput

The purpose of the second part of the simulation was to assess the network throughput and then to determine the levels, where the BPCP triggers MAC AM and Closed Network Mode. After preliminary tests and analysis, the optimal size of the Sampling Interval was set to 1s.

Experiments were performed under the assumption, that the network is dedicated to the VoIP service. Consequently, small size packets are transported across the network, see Table III. The CBR encoding was applied and the interval between packets was set to 20ms in the first case and to 40ms in the second. In both cases the number of VoIP streams was increased until the maximum network throughput was achieved. The collision index was measured as an additional parameter. This collision index is defined as the number of collided frames (coming from all stations) per the total number of frames sent across the network during the Sampling Interval.

<table>
<thead>
<tr>
<th>Case</th>
<th>UDP payload</th>
<th>Interval</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>40B, 80B, 160B</td>
<td>20ms</td>
</tr>
<tr>
<td>2</td>
<td>160B, 240B, 320B</td>
<td>40ms</td>
</tr>
</tbody>
</table>

The results of simulations are presented below. Fig. 6 and Fig. 7 show the network throughput $S$ vs. the traffic load $G$. These values were normalized to the data rate in the radio channel $R$, which was set to 2Mbit/s. The network throughput was measured by the BPCP component. Assuming the Poisson process, the traffic load was defined as follows:

$$G = G_{STA_1} + G_{STA_2} + ... + G_{STA_{N-1}} + G_{STA_N}$$

(3)

where $G_{STA_N}$ means the traffic load coming from Station number $N$. $G_{STA_N}$ can be calculated from the formula given below.

$$G_{STA_N} = (UDP_{Payload} + OH) \cdot n$$

(4)

$UDP_{Payload}$ is the product of the voice codec, e.g., 160B for G.711 codec, while $n$ denotes the number of frames sent per second.

$OH$ represents the total overhead and consists of the UDP, IP, MAC and PHY overheads [9].

$$OH = H_{UDP} + H_{IP} + H_{MAC} + H_{PHY}$$

(5)

For the purpose of analysis, the PHY overhead is defined as the sum of the following elements: DIFS, average size of Contention Window, PLCP header and Preamble.

$$H_{PHY} = DIFS + CW_{min} + H_{PLCP} + \text{preamble}$$

(6)
Typically, the bitrate of the G.711 voice codec is equal to 64kbit/s. The bitrate at the PHY layer increases up to 96.8kbit/s, when the overheads are taken into account.

Fig. 6 and Fig. 7 show the relation between the network throughput and the traffic load when packets are generated with the interval of 20ms and 40ms respectively. The shape of the curves is very characteristic. After linear growth of the network throughput the effect of saturation is observed for G/R equal to 0.5. If the traffic load exceeds 0.5 G/R, then the decrease of the throughput is observed, which means the decrease of the amount of data transferred across the network due to collisions and retransmissions. The delay and packet loss ratio exceeds the allowable level. Eventually, if the traffic load is still increasing, the network can be blocked completely. As a result, almost no data is transferred across the network.

If the packets interval increases, e.g., up to 40ms, the effect of saturation occurs for values greater than 0.5 G/R. When UDP Payload is set to 320B, the network is stable up to 0.75 G/R. If 160B UDP Payload is considered, the network throughput amounts to 0.65 for the traffic load reaching about 0.78 G/R.

Fig. 8 and Fig. 9 show the collision index vs. the normalized traffic load for the packets interval of 20ms and 40ms respectively. The value of collision index rises when the traffic load increases. The change of collision index is more rapid for packet of smaller size. For example, for the packets of 40B the collision index reaches the critical level 0.3 for 0.38 G/R. If 160B packets are considered, the critical level is observed for 0.5 G/R.

Higher traffic load causes sudden growth of collisions, which leads to network inefficiency.

The results presented in Fig. 6 - Fig. 9 are very consistent when comparing packets of the same size and packetization interval. The effect of the network saturation as well as the critical level of collisions is observed for the same value of the traffic load.

C. MAC Protocol States

At the beginning of the operation, special stations can cooperate with commercial and use standard access schemes, until BPCP detects insufficient bandwidth and initiates the MAC AM.

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). It is assumed, that the high rank users use special equipment with modified MAC parameters, while the low rank users are equipped with commercial devices operating with standard configuration, e.g., CW.

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. From this moment on, Wi-Fi network operates in a point-coordinated mode. Fig. 10 presents the states of MAC protocol for the proposed protocol extension.
An important issue is to determine the proper level of channel utilization for triggering between MAC AM and Closed Network Mode. To resolve this problem, the authors applied the Power optimization approach. Simulation results obtained for 2Mbit/s data rate and G.711 voice codec are presented below.

Fig. 11 shows the network throughput vs. traffic load. Triggering levels are denoted by A, B and C. If the throughput reaches limit denoted by B, the station switches from standard mode to AM. If it reaches another limit denoted by C, the station switches to Closed Network Mode. Fig. 12 presents collisions vs. traffic load. The critical level of collisions is denoted by A. This information may be used additionally by BPCP.

Results of simulations performed in order to estimate the acceptable number of VoIP connections, depending on the type of voice codec and MAC protocol parameters in a contention mode, were widely discussed in literature [10][11][20][21]. However, the question where and how to implement the AC mechanism and how to manage the traffic still remains open. The AC mechanism is necessary to prevent new calls if there is not enough bandwidth. In a contention mode, stations are not aware of the traffic load and try to transmit frames every time they have a packet to send. For this reason, a Closed Network Mode was proposed with a station named the Network Resource Manager (NRM) that manages the network.

IV. CLOSED NETWORK MODE

In this section, the Closed Network Mode was described. This mode enables the ad hoc network to self-organize and to determine the Resource Manager. Comparing to the PCF method, the infrastructure is not required, since the standard ad hoc node is selected to play the Resource Manager role. A set of new procedures was proposed and new frames were introduced as presented in next subsections.

A. Network Self-Organizing Mechanism

Network Self-Organizing Mechanism (NSOM) enables nodes to recognize the neighborhood and to determine the Resource Manager.

At the beginning, all stations work in a contention mode with standard parameters, Fig. 9. In the background, Neighbor Discovery Procedure is performed, which is based on broadcasting Neighbor Request and Neighbor Response frames [22]. This procedure allows recognition of the surroundings by collecting data from other nodes, namely: received signal strength and noise, battery level and rank of the station. Based on this information, each station determines its own NRM Readiness coefficient, which describes whether the station is ready to play a network manager role. This mechanism is still under implementation in OMNET++ v4.0.

If BPCP again detects the insufficient bandwidth coincidence, a station changes the mode to AM, while Neighbor Discovery Procedure is still in the background, Fig. 13.

When a first station detects the insufficient bandwidth, it initiates NSOM. Only stations with a certain NRM Readiness coefficient are allowed to participate in this phase.
If necessary, information on the network topology is refreshed by sending Neighbour Request frame, which contains the last NRM Readiness coefficient of the sending station, Fig. 14.

### B. Real-time Traffic Management

When the NRM station is determined, it sends a NRM Request broadcast frame informing that nodes are allowed to call for a bandwidth reservation, Fig. 15. Some stations respond with RT Confirm frames if they have RT packets to send, Fig. 16. The NRM Request frame is sent periodically to disseminate the list of queued stations and also the current queue limit. A more detailed description of the algorithm can be found in [19].

If the queue limit has reached or NRM Req Timeout has elapsed, the NRM station sends a RT Queue frame containing:
- queue size: number of STAs in queue,
- number of cycles: number of queue repetition,
- voice codec type,
- data rate,
- MAC address and order of stations in the queue.

![Figure 15. Channel reservation procedure.](image)

**Table IV.** Management Frames Size and Capacity

<table>
<thead>
<tr>
<th>Frame Type</th>
<th>Frame Max Size [B]</th>
<th>Number of Addresses</th>
</tr>
</thead>
<tbody>
<tr>
<td>NRM Request</td>
<td>240</td>
<td>35</td>
</tr>
<tr>
<td>RT Confirm</td>
<td>40</td>
<td>1</td>
</tr>
<tr>
<td>RT Queue</td>
<td>280</td>
<td>35</td>
</tr>
</tbody>
</table>

V. VOIP CAPACITY ANALYSIS

In order to assess the time required to organize the RT traffic, analytical investigations were performed. It was assumed that NRM is determined, avg. Backoff is equal to 100µs and 10 nodes compete for bandwidth reservation, Fig. 18.

![Figure 18. RT traffic scheduling procedure.](image)
If the data rate is set to 1Mbit/s, one cycle required to schedule the RT traffic takes approximately 14ms and this period reaches 10ms if the data rate increases to 2Mbit/s. Even if some level of collisions is assumed, this duration should not exceed 20ms.

Synchronous RT data transmission in a Closed Network Mode can be verified by using an analytical as well as simulation model. For the sake of convenience, e.g., in order to apply different input parameters, the authors used COMNET 3 simulation tool. The aim of simulations was to assess the channel utilization and the number of possible simultaneous VoIP calls as a function of the data rate. The following assumptions were made:

- network stations with commercial voice codec (G.711) with attributes defined in Table I,
- MAC/PHY parameters: SIFS = 10µs, DIFS = 50µs, PLCP Header + Preamble = 192µs,
- packets with standard protocol headers: MAC = 30B, IPv4 = 20B, UDP = 8B.

The channel utilization vs. the number of VoIP calls and various data rates was shown in Fig. 19 - Fig. 21.

In the phase of synchronous RT data transmission, there are only two cases when the channel is idle: DIFS, which precedes data frame transmission and SIFS between data frame and ACK frame.

An increasing number of VoIP connections leads to a linear growth of channel utilization, up to 90%. Better channel utilization is unachievable. This is a result of the fact that although the number of frames sent in a given period increases for higher data rates, there are still constant idle periods that separate frames.

Table VI summarizes the results of experiments showing the maximum number of voice streams sent across the network when synchronous transfer is applied.

<table>
<thead>
<tr>
<th>Codec</th>
<th>Data rate [Mbit/s]</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>8 13 22 27</td>
</tr>
<tr>
<td>G.726</td>
<td>11 17 25 28</td>
</tr>
<tr>
<td>G.728A</td>
<td>20 28 40 46</td>
</tr>
</tbody>
</table>

Figure 19. Channel utilization vs. number of VoIP streams for G.711 voice codec (64kbit/s).

Figure 20. Channel utilization vs. number of VoIP streams for G.726 voice codec (32kbit/s).

Figure 21. Channel utilization vs. number of VoIP streams for G.728A voice codec (16kbit/s).

The delay of RT packets results from the data rate and the sequence number of a given station in the whole queue. Thus, this delay does not exceed two dozens of milliseconds. When the data rate grows, the time needed for transmission of one frame becomes shorter, while DIFS and SIFS remain on the same level. Therefore it is possible to set up more VoIP connections, however the channel utilization cannot exceed 90%. When G.711 codec is used and the data rate is set up to 11Mbit/s, up to 27 VoIP calls are available.

### Table V. AVERAGE SIZE AND NUMBER OF EXCHANGED FRAMES

<table>
<thead>
<tr>
<th>Frame type</th>
<th>Frame average size [B]</th>
<th>Frames number</th>
</tr>
</thead>
<tbody>
<tr>
<td>NRM Request</td>
<td>100</td>
<td>5</td>
</tr>
<tr>
<td>RT Confirm</td>
<td>40</td>
<td>10</td>
</tr>
<tr>
<td>RT Queue</td>
<td>100</td>
<td>1</td>
</tr>
</tbody>
</table>

### Table VI. NUMBER OF VOICE STREAMS

<table>
<thead>
<tr>
<th>Codec</th>
<th>Data rate [Mbit/s]</th>
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<td>11 17 25 28</td>
</tr>
<tr>
<td>G.728A</td>
<td>20 28 40 46</td>
</tr>
</tbody>
</table>
VI. CONCLUSIONS

We have presented the concept of the resource management strategy in ad-hoc networks for rescue operations. This strategy is a result of the new outlook on the 802.11 WLANS capabilities and performance enhancement.

A set of novel procedures was developed with a view of organizing the network and managing the real-time traffic. These procedures were validated analytically and by simulations, and results were included. The proposed method of the available bandwidth measurement and estimation works correctly. The triggering levels were determined to switch the MAC protocol mode operation: Standard, MAC Acquisitive Mode or Close Network Mode.

The procedure of RT data synchronous transmission in a Closed Network Mode was verified by simulation. Results of tests allowed estimating the channel utilization achieving over 80% when synchronous transmission was applied. If the number of stations in a queue is set correctly, the delay of the RT data frame transmission is limited to two dozens of milliseconds and is determined mainly by the data rate.

The proposed mechanisms were developed as a result of a completely new approach to the support of RT data transmission in 802.11 ad-hoc network. They enrich standard procedures and enable an efficient utilization of the channel.

VII. FUTURE WORK

In this article, we have only presented the resource management strategy to support VoIP traffic. We described the procedures enabling the organization of network and real-time traffic management.

The presented results were obtained under the assumption that only UDP traffic is transferred across the network. 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 are going to investigate how to efficiently manage the network where VoIP streams are combined with the TCP flows.

The issue of nodes mobility is crucial for NRM determination and will be the topic of further study.

Furthermore, we would like to devote attention to the aspect of the distributed network management. This includes optimization of the scheme for determining the secondary resource manager when the first manager terminates unpredictably.

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