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

Janusz Romanik
Radiocommunications Department
Military Communications Institute
Zegrze, Poland
j.romanik@wil.waw.pl

Piotr Gajewski, Jacek Jarmakiewicz
Faculty of Electronics
Military University of Technology
Warsaw, Poland
{pgajewski, jjarmakiewicz}@wel.wat.edu.pl

***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 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.

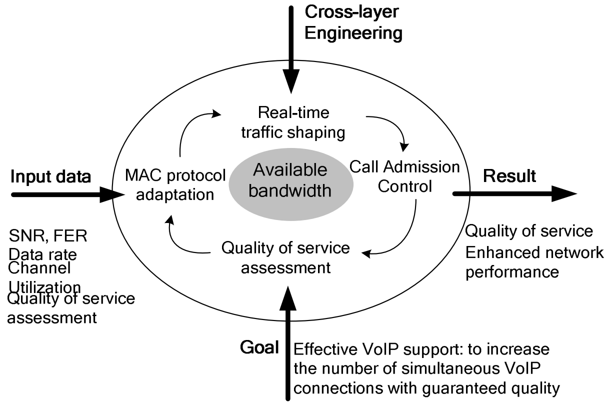


Figure 1. Performance enhancement of WLAN for VoIP support. [1]

The network model assumes WLAN based solution. The authors decided to use the IEEE 802.11b standard as offering good throughput and modulations more resistant to interferences, which is a real advantage of the network operating in ad hoc mode. MAC QoS mechanisms defined in the IEEE 802.11e standard were also taken into account. These mechanisms are a good starting point to enable prioritization and bandwidth reservation in ad hoc network [13][14]. In particular, the authors introduced the adaptation of a CW size to the type of frame and the rank of the user.

Another issue concerns the optimal balance between the traffic load and the services quality in ad-hoc networks.

It is expected that the proposed range of adaptation and introducing of new mechanisms will not demand a high cost of implementation and will be feasible.

Fig. 1 illustrates the concept of efficiency improvement of WLAN 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 real-time traffic shaping or MAC adaptation if the available bandwidth is too small or if the level of service is not satisfactory. The CAC mechanism prevents new VoIP calls if 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 commercial users. It perfectly corresponds to the scenario involving humanitarian aid operations, when volunteers help people in service. Another example can be the 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 how to assess the resources of the network, e.g., channel utilization. Since ad-hoc networks are bandwidth limited, not all measurement methods can be applied [16][17]. From that reason, non-intrusive methods are proposed for ad hoc networks [18].

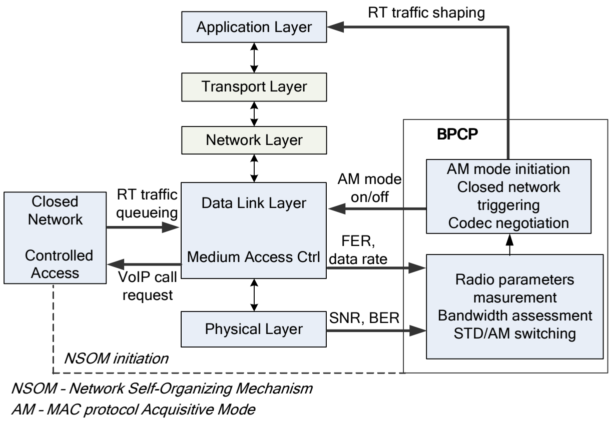


Figure 2. BPCP alignment with protocol stack.

Fig. 2 illustrates the proposed approach on the basis of the extended protocol stack with cross-layer interactions. Bandwidth Prediction Control Protocol (BPCP) allows monitoring of parameters in the physical layer, to measure the channel utilization level and also to switch MAC protocol states, as explained in subsequent sections. Real-time (RT) traffic shaping relays on codec negotiation and audio packets aggregation. Closed Network Mode is based on the concept of the Resource Manager that controls traffic in the network.

BPCP, MAC Acquisitive Mode (AM) and RT traffic shaping mechanisms do not enforce changes to IEEE 802.11 standard. These new mechanisms only enhance functionality of the station that still operates in standard way. Closed Network Mode requires completely new procedures and set of extra frames exchanged in order to organize the network and the RT queue. Although stations operate according to non-standard scheme, this mode offers a real advantage of better channel utilization, which results in increased number of voice streams [26].

In the proposed solution, BPCP provides CAC mechanism with the knowledge of the current and expected channel utilization. These mechanisms exchange the data thanks to the cross-layer engineering, as the CAC operates outside standard WLAN sublayer.

## A. Bandwidth Estimation

The available bandwidth is crucial for optimization of the Wi-Fi ad-hoc network. Therefore, the authors proposed to implement BPCP that enables to measure the channel utilization level and to estimate the available bandwidth.

BPCP takes advantage of WLAN card drivers that enable the measurement of signal to noise ratio (SNR) in the PHY layer and bit error rate (BER) calculation, and passing these parameters to the Data Link Layer. If nodes operate in promiscuous mode, they can receive all the traffic sent across the network. As a result, the bandwidth utilization is assessed in all nodes of the network independently and continuously for predefined periods called Sampling Intervals.

From the PHY layer point of view, stations can detect the channel state (idle - no transmission or busy - transmission) 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).

$$t_{AF} = \frac{AF\_length\,(bits)}{AF\_bitRate\,(bits/\sec)} \tag{1}$$

Having knowledge of $t_{AF}$ parameters, it is then possible to determine the channel utilization coefficient for the interval, e.g., from $t_1$ to $t_2$

$$U = \frac{t_{d1} + t_{d2} + ... + t_{dn} + n \cdot t_{ACK} + n \cdot SIFS + n \cdot DIFS}{t_2 - t_1} \tag{2}$$

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

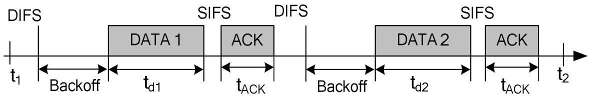


Figure 3. Transmission scheme in a contention mode of WLAN.

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.

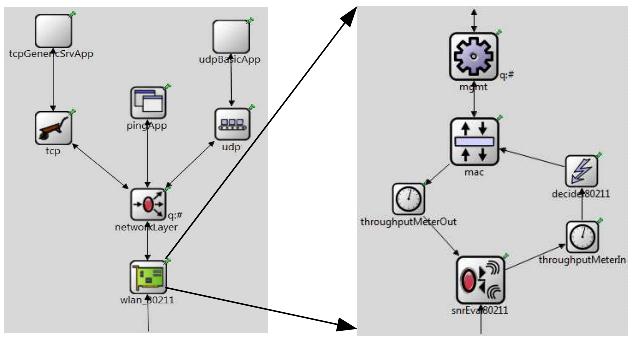


Figure 4. The extended WLAN sublayer of the mobile node.

TABLE I. MAC PARAMETERS

| Parameter | Value |
|---|---|
| DIFS | 50 µs |
| SIFS | 10 µs |
| Slot Time | 20 µs |
| CWmin | 32 |
| CWmax | 1023 |
| Data Rate | 2Mbit/s |
| PHY header | 192 µs |
| MAC header | 34 bytes |
| ACK | 304 µs |

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

| Codec | G.711 |
|---|---|
| Bit rate [kbit/s] | 64 |
| Framing interval [ms] | 20 |
| Payload [B] | 160 |
| Packets/sec | 50 |

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 1,3Mb/s, Fig. 5. Higher traffic load will cause an increase of the collision rate and a drop in network efficiency.

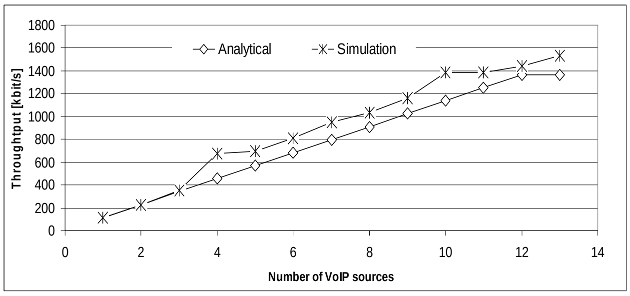


Figure 5. Wi-Fi network throughput - contention mode, data rate 2Mbit/s.

## 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 III. SIZE OF PACKETS

| Case | UDP payload | Interval |
|---|---|---|
| 1 | 40B, 80B, 160B | 20ms |
| 2 | 160B, 240B, 320B | 40ms |

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:

$$G = G_{STA\_1} + G_{STA\_2} + ... + G_{STA\_N-1} + G_{STA\_N} \quad (3)$$

where $G_{STA\_N}$ means the traffic load coming from Station number N.

$G_{STA\_N}$ can be calculated from the formula given below.

$$G_{STA\ N} = (UDP\_Payload + OH) \cdot n \quad (4)$$

*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].

$$OH = H_{UDP} + H_{IP} + H_{MAC} + H_{PHY} \quad (5)$$

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*.

$$H_{PHY} = DIFS + CW_{Avg} + H_{PLCP} + preamble \quad (6)$$

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 shows 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 0,6 G/R.

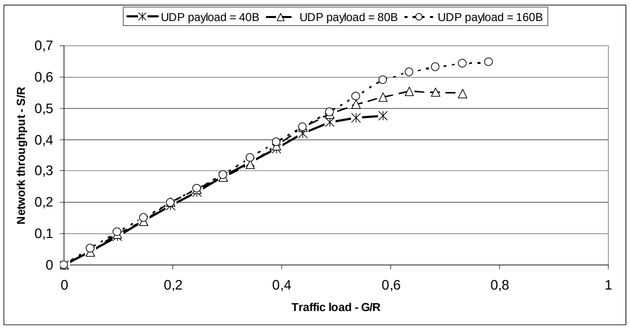


Figure 6. Normalized network throughput – packets generated with the interval of 20ms.

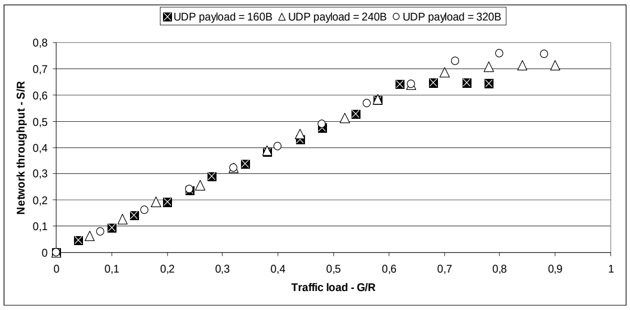


Figure 7. Normalized network throughput – packets generated with the interval of 40ms.

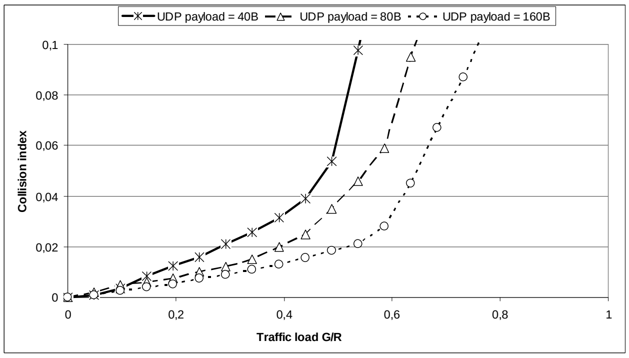


Figure 8. Collision index – packets generated with the interval of 20ms.

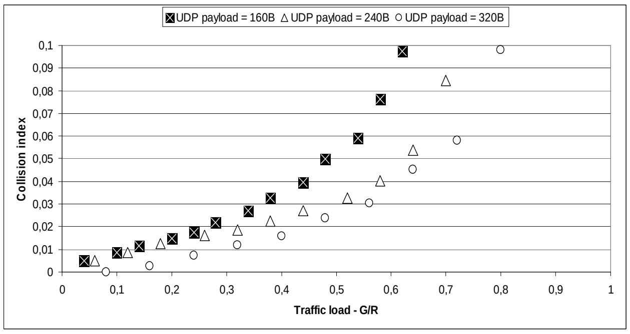


Figure 9. Collision index – packets generated with the interval of 40ms.

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.

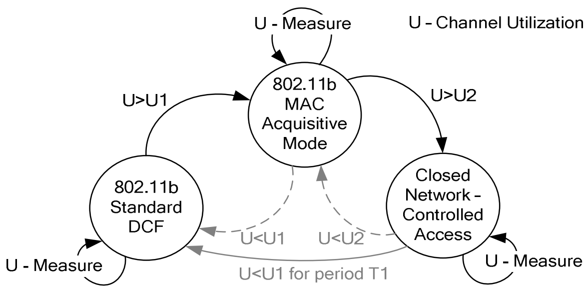


Figure 10. MAC protocol states.

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 Pareto 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.

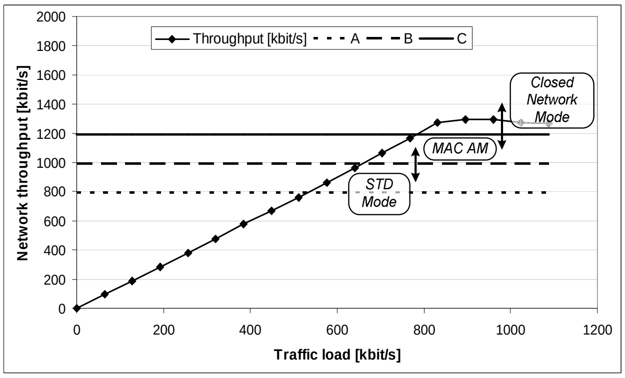


Figure 11. Network throughput vs. traffic load (where triggering levels are denoted as follows: A - switch from AM Mode to std., B - switch from std. to AM Mode, C - switch to Closed Network Mode).

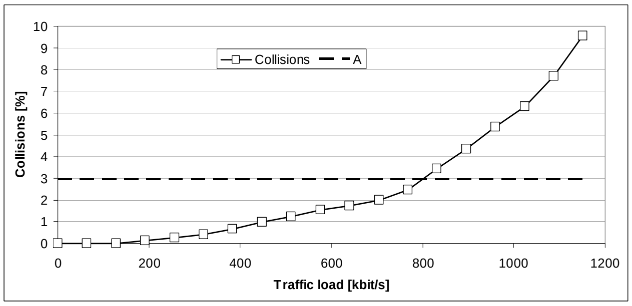


Figure 12. Collisions vs. traffic load (where A denotes the critical level of collisions).

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.

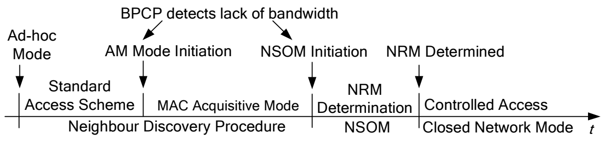


Figure 13. Network Self Organizing procedure.

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.

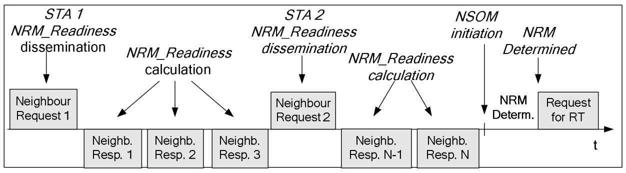


Figure 14. Neighbour Discovery procedure.

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].

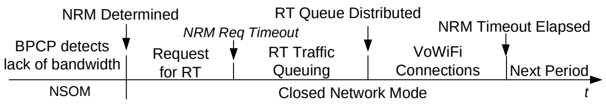


Figure 15. Channel reservation procedure.

If the queue limit is 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.

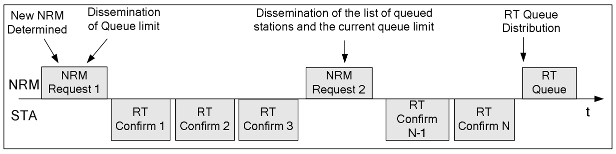


Figure 16. RT traffic management.

After receiving the RT Queue frame, the first station on the list is allowed to transmit after DIFS and receives an ACK frame after SIFS, Fig. 17. The next station in queue transmits data frame after DIFS. The number of cycles describes how long nodes will transmit data in a given order.

After each transmission of DATA and ACK, stations decrease their *TransmissionIndex* and are allowed to send after it reaches zero. After a predefined number of cycles, the NRM station again sends a NRM Request frame to give a chance to transmit for stations that were out of queue during the preceding period.

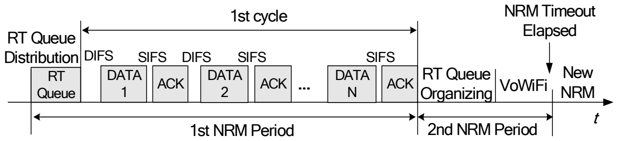


Figure 17. RT traffic queue.

An unpredictable NRM termination may occur, e.g., as a result of depletion of the battery, which should be taken into account. In such a situation nodes will detect a lack of frames from NRM for the assumed timeout. Since this moment on, the station with the second highest *NRM Readiness* coefficient starts playing this role.

In order to organize a closed network and manage RT traffic, the following management frames were introduced:

- Neighbor Request and Neighbor Response - for neighborhood discovering,
- NRM Request - for initiation of the RT traffic queuing phase,
- RT Confirm - for the bandwidth reservation,
- RT Queue - distribution of RT traffic queue.

The detailed description of the management frames structure can be found in [19]. Because all of these frames are of a broadcast type, all receiving stations are forced to process it in the data link layer, although acknowledgement is not sent. The structure of new frames is the same as defined in the IEEE 802.11 standard for management frames and consists of MAC Header and Frame Body containing information fields. The maximum size and capacity of frames are presented in Table IV.

TABLE IV. MANAGEMENT FRAMES SIZE AND CAPACITY

| **Frame Type** | **Frame max size [B]** | **Number of addresses** |
|---|---|---|
| NRM Request | 240 | 35 |
| RT Confirm | 40 | 1 |
| RT Queue | 280 | 35 |

## 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.

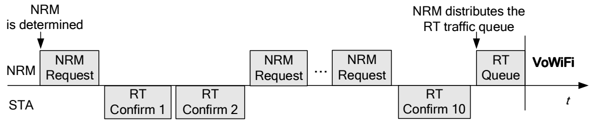


Figure 18. RT traffic scheduling procedure.

The size and the amount of frames exchanged in this procedure are presented in Table V.

TABLE V. AVERAGE SIZE AND NUMBER OF EXCHANGED FRAMES

| Frame type | Frame average size [B] | Frames number |
|---|---|---|
| NRM Request | 100 | 5 |
| RT Confirm | 40 | 10 |
| RT Queue | 100 | 1 |

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.

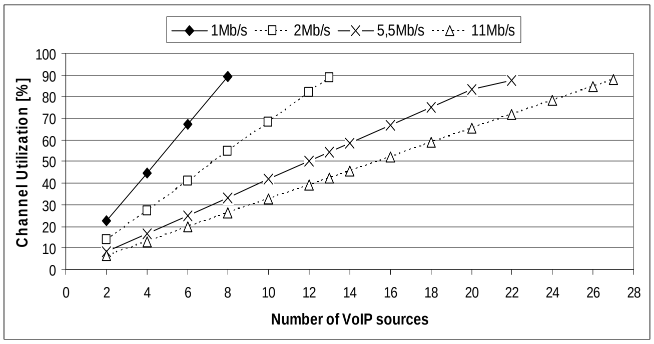


Figure 19. Channel utilization vs. number of VoIP streams for G.711 voice codec (64kbit/s).

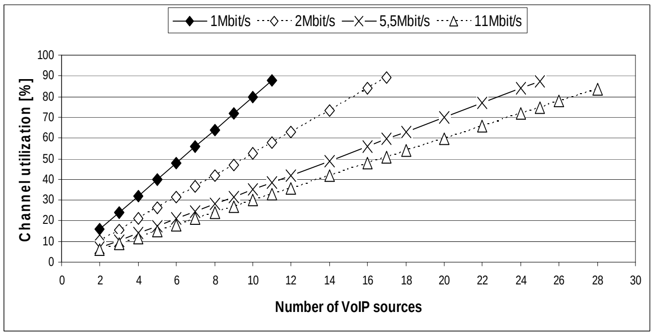


Figure 20. Channel utilization vs. number of VoIP streams for G.726 voice codec (32kbit/s).

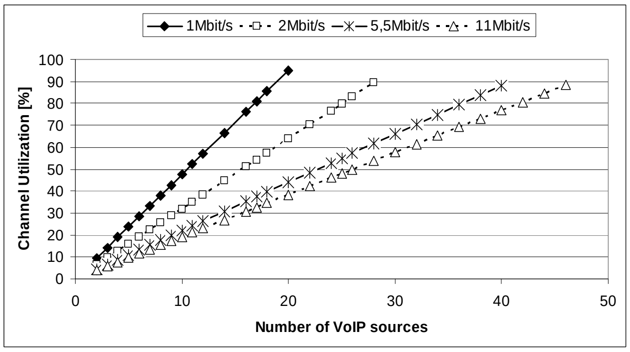


Figure 21. Channel utilization vs. number of VoIP streams for G.728A voice codec (16kbit/s).

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 11Mb/s, up to 27 VoIP calls are available.

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

TABLE VI. NUMBER OF VOICE STREAMS

| Codec | Data rate [Mbit/s] | | | |
|---|---|---|---|---|
| | 1 | 2 | 5,5 | 11 |
| G.711 | 8 | 13 | 22 | 27 |
| G.726 | 11 | 17 | 25 | 28 |
| G.728A | 20 | 28 | 40 | 46 |

## 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.

## ACKNOWLEDGEMENT

This work was supported by the Polish Ministry of Science and Higher Education under grant number 6266/B/T00/2010/39.

## REFERENCES

[1] J. Romanik, P. Gajewski, and J. Jarmakiewicz “A Resource Management Strategy to Support VoIP across Ad hoc IEEE 802.11 Networks,” The Fourth International Conference on Communication Theory, Reliability and Quality of Service - CTRQ, Budapest, Hungary, April 17-22.2011

[2] A. F. da Conceicao, J. Li, D. A. Florencio, and F. Kon “Is IEEE 802.11 ready for VoIP,” IEEE Workshop on Multimedia Singal Processing, October 2006, pp. 108-113

[3] T. Maseng, “Wireless Tactical Local Area Network,” Multinational CDE, Pre-Symposium Workshop, Sundvollen, Oslo, Norway 2002

[4] J. Lopatka and R. Krawczak, “Military Wireless LAN Based on IEEE 802.11b Standard”, in Military Communications, Meeting Poceedings RTO-MP-IST-054, Poster 2, France 2006, pp. 21-28

[5] S. Srivathsan, N. Balakrishnan, and S. S. Iyengar, Guide to Wireless Mesh Networks - Scalability in Wireless Mesh Networks,Springer 2009

[6] S. Choi and J. Yu, QoS Provisioning in IEEE 802.11 WLAN, John Wiley & Sons, Inc,2006.

[7] H. Yoon, J. W. Kim, and D. Y. Shin, “Dynamic Admission Control in IEEE 802.11e EDCA-based Wireless Home Network,” IEEE Consumer Communication and Networking Conference, January 2006, pp. 1-5

[8] P802.11e – Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications. Amendment 8: Medium Access Control (MAC) Quality of Service Enhancements, Nov. 2005

[9] W. Wang and S. C. Liew, “Solutions to Performance Problems in VoIP over 802.11 Wireless LAN,” IEEE Transaction on Vehicular Technology, 54(1), 2005, pp. 336-384

[10] N. Hegde, A. Proutiere, and J. Roberts, “Evaluating the Voice Capacity of 802.11 WLAN under Distributed Control,” In Proc of LanMan, 2005, pp. 1-6

[11] L. Gannoune, “A Comparative Study of Dynamic Adaptation Algorithms for Enhanced Services Differentiation in IEEE 802.11 Wireless Ad-Hoc Networks,” Proc. of AICT/ICIW, February 2006

[12] L. Cai, Y. Xiao, X. Shen, and L. Cai “VoIP over WLAN: Voice capacity, admission control, QoS and MAC,” International Journal of Communication Systems, 2006, (19) pp. 491-508

[13] J. Liu and Z. Niu, “A Dynamic Admission Control Scheme for QoS Supporting in IEEE 802.11e EDCA,” Proc. WCNC 2007, pp. 3697-3702

[14] P. Wang, H. Jiang, and W. Zhuang, “A New MAC Scheme Supporting Voice/Data Traffic in Wireless Ad Hoc Networks,” IEEE Transactions on Mobile Computing, vol. 7, no. 12, December 2008, pp. 1491-1503

[15] J. Z. Haas and J. Deng, “Dual Busy Tone Multiple Access (DBTMA) - A Multiple Access Control Scheme for Ad Hoc Networks,” IEEE Trans. Comm., vol. 50, no.6, June 2002, pp. 975-985,

[16] R. Prasad, M. Murray, C. Dovrolis, and K. Claffy, “Bandwidth Estimation: Metrics, Measurement Techniques, and Tools,” IEEE Network, vol. 17, no. 6, pp. 27-35, 2003

[17] G. Chelius and I. G. Lassous, “Bandwidth Estimation for IEEE 802.11-Based Ad Hoc Networks,” IEEE Transactions on Mobile Computing, vol. 7, no. 10, 2008, pp. 1228-1241

[18] C. Sarr, C. Chaudet, G. Chelius, and I. G. Lassous, “A node-based available bandwidth evaluation in IEEE 802.11 ad-hoc networks,” International Journal of Parallel, Emergent and Distributed Systems, vol. 21, 2006, pp. 423-440

[19] J. Romanik, P. Gajewski, and J. Jarmakiewicz, “Performance enhancement of Wi-Fi ad-hoc network for VoIP support,” Proc. of Military Communications and Information Systems Conference, Wroclaw, 2010, pp. 525-535

[20] S. Garg and M. Kappes, “An experimental study of throughput for UDP and VoIP traffic in IEEE 802.11b networks,” WCNC, March 2003, pp. 1748-53,

[21] F. Anjum, et al., “Voice performance in WLAN networks - an experimental study,” Proc. IEEE Globecom, 2003, pp. 3504-3508

[22] M. Bednarczyk and M. Amanowicz, “Wireless Relay Control Protocol (WRCP),” Proc. of Military Communications and Information Systems Conference, Bonn, Germany, September 2007, ISBN 978-3-934401-16-7

[23] H. Zhang, J. Zhao, and O. Yang, “Adaptive Rate Control for VoIP in Wireless Ad Hoc Networks,” Proc. of IEEE International Conference on Communications, 2008, pp. 3166-3170

[24] J. Barcelo, B. Ballalta, and C. Cano, “VoIP Packet Delay in Single-Hop IEEE 802.11 Networks,”WONS 2008, Barcelona, pp. 77-80

[25] S. Sangho and H. Schulzrinne, “Measurement and Analysis of the VoIP Capacity in IEEE 802.11 WLAN,” IEEE Transactions on Mobile Computing, vol. 8, 2009, pp. 1265-1279

[26] J. Romanik, P. Gajewski, and J. Jarmakiewicz, “An adaptive MAC scheme to support VoIP across ad hoc IEEE 802.11 WLANs,” Military Communications and Information Systems Conference, Amsterdam, 17-18.10.2011, in: Military Communications and Information Technology: A Comprehensive Approach Enabler, ISBN 978-83-62954-20-9, pp. 496-505.