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Adaptive Mechanism for Aggregation with fragments retransmission in high-speed wireless networks

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Abstract

The aim of this paper is to propose an adaptive mechanism for aggregation with fragments retransmission, to create appropriate simulator and to examine performance of this aggregation mechanism, taking into account time-varying radio channel characteristics and their strong relation with errors.

Keywords*: Medium access control (MAC), physical layer (PHY),* high *efficiency, IEEE 802.11n, IEEE802.11, wireless LAN (WLAN)* .

1 Introduction

Modern wireless computer networks offer more high-speed data transmission in the physical layer (PHY) and using highly efficient protocols in the Medium Access Control layer (MAC) to access the communication medium.

High-speed of physical layer does not lead directly to increased efficiency of the MAC layer. The reason is that increasing speed leads to faster transmission of the MAC part in frame (user data), but the transmission time of PHY header and the back off time (to avoiding collisions of several simultaneously transmitting stations) has not decreased substantially. For example, the new 802.11n standard offers speeds up to 600 Mbps. Transmission time of PHY header, however, is 48 µs. The maximum size of frame is limited to 7955 B. Thus, at a speed 150 Mbps, the time for transmitting user data is 424 µs, which means the proportion of transmission time for PHY header in frame is more than 10%. It is known that even under the best conditions, the efficiency of MAC layer

(MAC_Layer_Speed/PHY_Layer_Speed) in 802.11n fall from 42% at a speed of 54Mbps to only 10% at speed of 432Mbps [9].

Appropriate solution to overcome this phenomenon in high-speed wireless networks is the use of mechanisms for aggregating packets.

The known in literature analytical and simulation models can't reflect some behaviors of the mechanisms for aggregating packets and limitations of Internet traffic. Most studies of mechanisms for aggregating packets using models in which traffic has a Poisson or Bernoulli distribution. These models cannot capture the strong correlative nature of actual network traffic and the sequence of lost packets (due to error prone radio- channel).

In this paper an adaptive mechanism for aggregation with fragments retransmission is proposed, taking into account time-varying radio channel characteristics and their strong relation with errors.

2 Modified mechanism for aggregation with fragments retransmission

In the aggregation mechanism with fragment retransmission -AFR, multiple packets are aggregated into one large frame to be sent. Using technology of fragmentation, whereby if the packets are larger than a threshold, they are split into fragments that are re-transmitted in case of loss, rather than retransmitting of whole aggregated frames.

The following proposed adaptive mechanism for aggregation with fragments retransmission A-AFR is a modification of AFR, which solve two major problems:

The first one is to improve delays in AFR aggregation. In the literature [2] is proposed aggregation to do with utilization above a certain threshold:

$$\rho^* = \frac{1}{4}(7 - 3\sqrt{1+8h}) \quad (1)$$

where ***h*** *is the proportion of transmission time for the header (MAC and PHY) components from the total time for transmitting frame.*

In low intensity of arrival packets in the buffer, respectively utilization ($\rho = \lambda/\mu$) below this threshold, aggregation is not done, and each new arrival packet forms a frame.

The second problem is related to the losses of frames in wireless networks and optimizing length of the fragments that form the aggregated frame and a coefficient of efficiency is used as a criterion for optimality. In a previous publication [9] has proposed a formula by which to determine the optimum length of the fragments, which balance between additional service information to be transmitted (the headers of the fragments) and the information surplus retransmission (retransmission of whole fragment even if it is only one wrong bit) after losses. The optimal length of the fragments that form the aggregated frame is:

$$d = -c + \sqrt{cT} \quad (2)$$

where ***c***- *is length of header* and ***d*** – *is length of data field in fragments*.

In A-AFR and AFR mechanisms, the MAC frame consists of a header and body (Fig. 1). All fields of the MAC header remain unchanged, only three new fields added - size of fragment, number of fragment and reserved field.
The body of frame contains the headers of fragments and the bodies of fragments and control field for checking the corresponding fragment (FCS-Fragment Check Sequences). Each header fragment consists of six fields: ID of the packet (PID), length of the package (pLEN), start position (startPos), offset field (offset), reserved for future use fields and FCS. StartPos is used to indicate the position of body fragment in the frame and offset (offset) is used to record the position of this fragment in the packet.

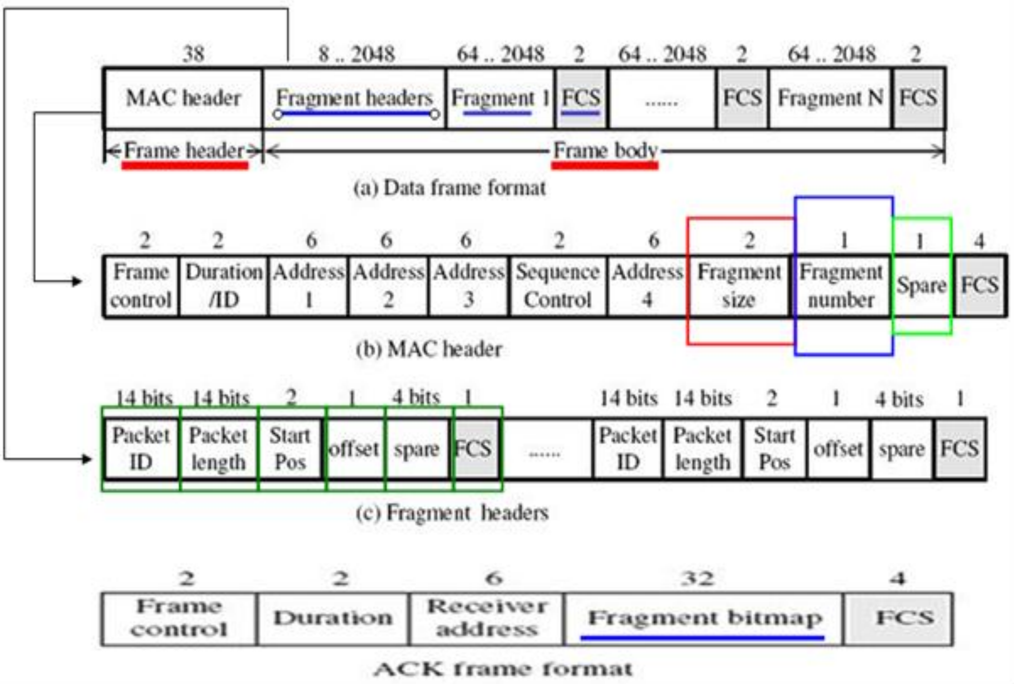

Fig.1 A-AFR Frame formats

3 Simulation model

The A-AFR mechanism and more precisely the transmitter assigns unique identifier (ID) to each fragment in the aggregated frame (Fig.1). In the receiver side fragments of a packet are concatenated according to their IDs. After the transmission of aggregated frame, constituent fragments temporarily buffered while the acknowledgement frame (ACK frame) arrives back to the transmitter. This happens with some delay (see feedback in Fig. 2). In case that a positive acknowledgment (ACK) arrives for given fragment, this fragment will be removed from the retransmission buffer (waiting buffer). If the acknowledgement arrives negative (NACK) – the fragment will be transmitted again.

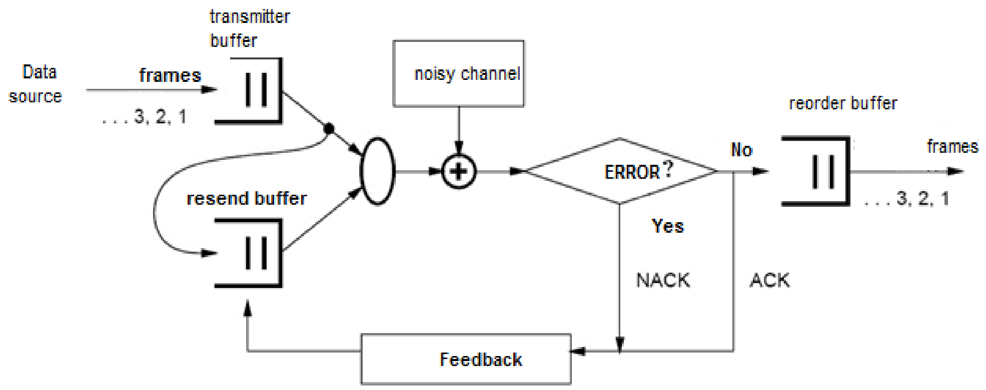

Fig.2. Packets transmission over wireless network

Although the transmitter sends fragments of packets in the correct order, the order of the fragments into the receiver may be broken due to the occurrence of errors, respectively retransmission. So correctly received fragments have to wait in a buffer until the lost fragments (with the missing IDs) are received correctly. The buffer for re/sequencing is located in the receiver. Once all fragments of an aggregated frame arrive, they are released from resequencing buffer (in the correct order) and packets are forwarded to the upper levels.

Fig. 3 shows the various delays that fragments of packets undergo in their transmission across a wireless network using the A-AFR protocol. Total delay- Tt is the delay for packets transport, including the delivery delay- Td and the delay in the transmitter queue also called queuing delay- Tq. The delay Tt is the time elapsed since the first transmission (of fragments) of the packet until the moment in which this packet leaves resequencing buffer. The delay in the transmitter queue (Tq) is defined as the elapsed time from the arrival of packets in the buffer to the first attempt of transmission. Delivery delay (Td) includes delay of retransmission and delay of rearranging. The retransmission delay is defined as the time elapsed since the first transmission of the fragment until it successfully arrives at the receiver. Delay for rearrangement of packet fragments (resequencing delay) is equal to the time that the packet waits until all its fragments arrive in resequencing buffer.

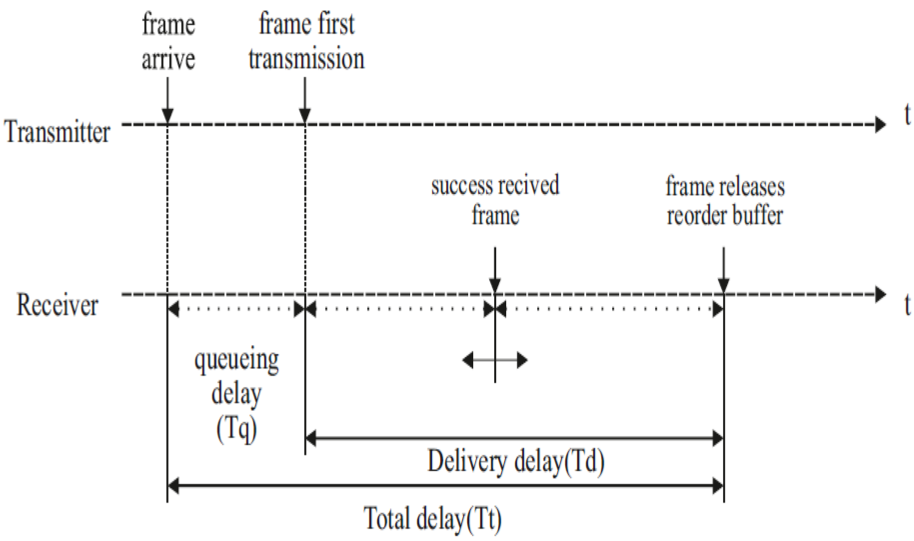

Fig.3. Timeline of transmission process

The generation of the input stream in the model is achieved by ON-OFF process. Modeling of time-varying radio channel is also used ON-OFF process.

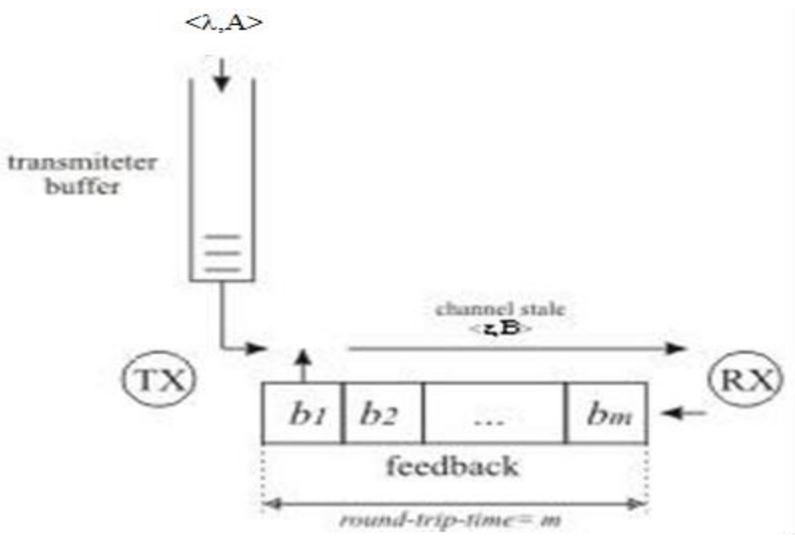

Fig.4. Queuing system

In the queuing system we assume that a fragment is transmitted per slot (the time in model is slotted). The time in which an information comes for the status of the fragments of an aggregated frame (round-trip-time) is equal to m slot (Fig. 4), where $m>1$. This means that the available packets in the buffer are fragmented, aggregated and transmitted, but will not leave the queuing system before waiting at least m slots.

The arriving of fragments of packets in the buffer of transmitter is described by:

- Intensity of the arrival packets - λ.

- Average number of packets in one aggregated frame –p.

- Average number of 128B fragments in a packet - L.

The last parameter characterizes the process of fragmentation of packets, i.e. the average number of fragments aggregated in one frame is A = p.L.

New arrival packet is immediately transmitted [9] only when the buffer is empty as well as there is no request for retransmission of fragment/s of earlier transmitted packet. This is because retransmission of the fragments has a higher priority than fragmentation, aggregation and transmission of newly arrived packets.

The sent data from the transmitter reaches the receiver on radio channel in which there is interference which can lead to loss of fragments. In the proposed model this error prone channel is also modeled by ON-OFF generator (process with two states) and is described by parameters:

- Error probability of the channel also called channel error probability- ε;

- Average error burst length (length of the sequence of lost fragments due to the packet error)- B.

The receiver responds with a positive or negative acknowledgment (ACK / NACK) depending on whether the fragment was received without errors or with errors. After the round-trip-time, i.e. after m slots the transmitter gets feedback (ACK/ NACK) and then starts transmission of a new aggregated frame or retransmission of lost fragment/s (for which is arrived a negative acknowledgement - NACK). Corresponding flags - bi $(i = 1, 2, ... m)$ are used to model the result of transmission in mi slot, where $bi = 1$ - means that the transmission of *i*-th fragment is not successfully and its retransmission is necessary, otherwise i.e. $bi = 0$ and the transmission is successfully.

In the proposed model is assumed that no errors occur in transmission of acknowledgements (ACK / NACK), i.e. all acknowledgements arrive to the transmitter.

General Purpose Simulation System (GPSS) has been chosen to create simulator for evolution of the A-AFR mechanism. The proposed modeling approach reflects the requirements and limitations of the GPSS language environment and accurately describes the parameters and the processes in the wireless network.

4 Simulation results

The duration of simulations is 100 000 packets, each with an average length L= 3.2 fragments (according statistics [9] for traffic in the Internet) and transfer rate λ = 150 Mbps. The values of all delays are converted to microseconds.

The behavior of delays is examined versus the varying average number of fragments in aggregated frame (A), at different intensities of incoming packets ($\rho = \lambda / \mu = 0.4$ and $\rho = 0.6$). The average errors burst length in this case is chosen to be B = 3 fragments. Figure 5 shows the delay in the queue Tq and the delivery delay Td for the given above parameters values. As one can see the delivery delay of packets-Td does not change significantly when ρ and A change. The delay in the queue (of the transmitter) is increased with increasing ρ, as it is expected. The graph also shows that the delay in the queue increases with increasing parameter A. This can be explained by the fact that packets fragments arrived explosively (burst) and accumulate in the queue, which increases the value of Tq.

The above means that this delay, and thus total delay can be large even at low intensity, but explosively generated fragments.

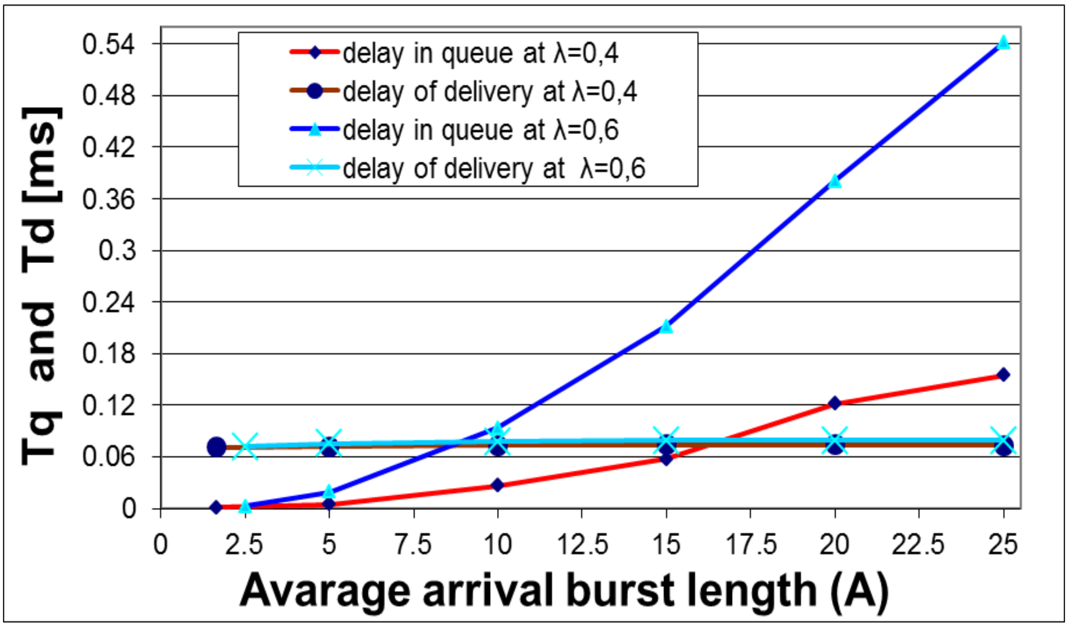

Fig.5. Average delay in queue and delivery delay, for m = 10, = 0.1, B = 3 as a function of A, with values of ρ = 0.4 and 0.6.

The results for the delays in the transmitter queue are compared with these calculated by well-known formula $W = 1/(\mu-\lambda)$, and as expected the differences are not greater than 20%, which is a kind of verification of proposed model.

Fig.6 shows the delivery delay as a function of load -ρ (respectively, the intensity of the packets arrival) at ***m* = *10*, = *0.1*, *A* = *2.5***, and ***B* = *3*, *10*, *60***. Can be expected that with increasing ρ, the delay will be increased because the system becomes more and more loaded. This is correct for the queue delay, but it is not true for the delivery delay. In fact, when B is close in value to the number of slots for receiving feedback -***m***, i.e. channel is correlated; the delivery delay hardly depends on the intensity of packets arrival and may even decrease with increasing ρ. This can be explained by the fact that when the channel is highly correlated it is possible to have a long series of slots, where the channel is in "good" condition. The above phenomenon is more pronounced for large values of error burst length B.

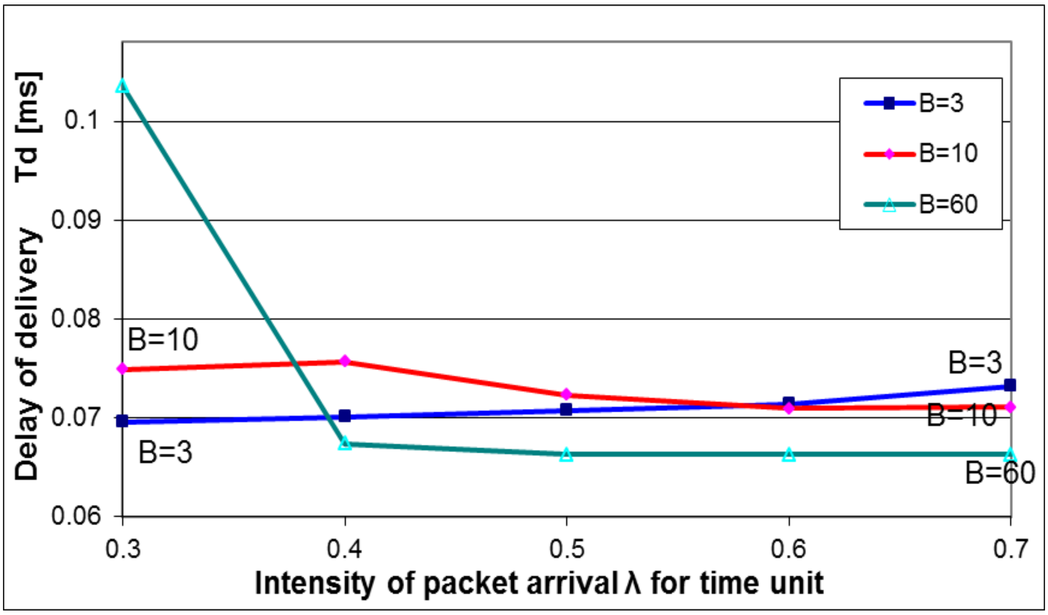

Fig.6. Averages for delivery delay as a function of ρ, where m = 10, = 0.1, A = 2.5, at different values for B.

Fig.7 shows the total delay as a function of ***B***, where ρ ***= 0.6***, ***A = 7***, ***m = 10*** and ***m = 0.1***. As seen from the graphs the total delay initially decreases and then increases.

The reason for this is that from one side at small and medium error burst length, ***B***, the delay of retransmission dominated (see Td and Tq). From other side, by increasing B, the delay of retransmission reduces and hence also reduces the total delay. For large values of B, the delay in the queue has a strong character and the total delay increases (see Tq).

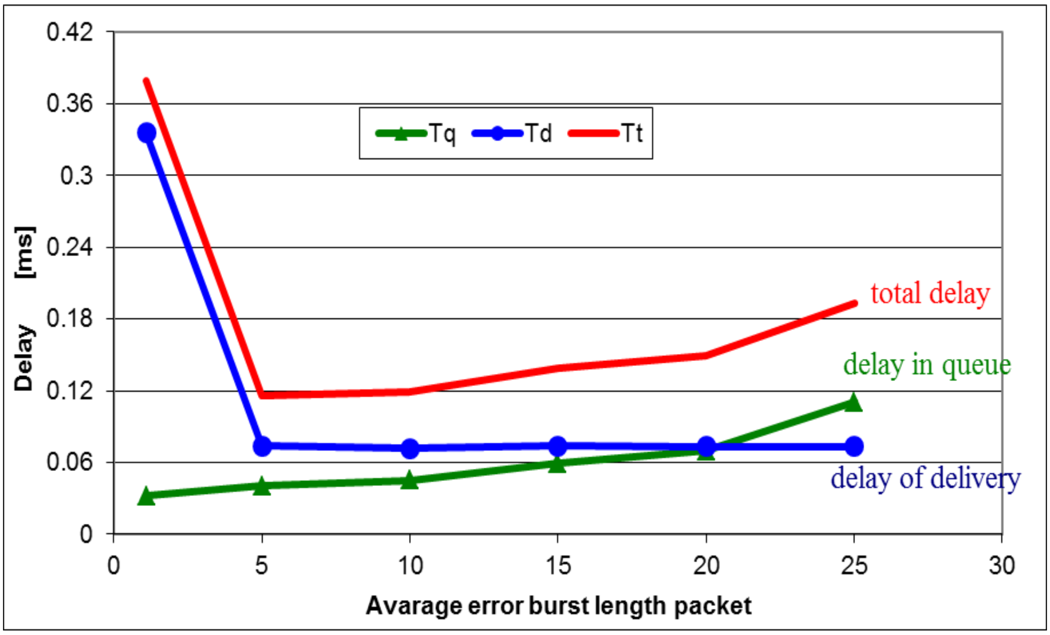

Fig.7. The total delay as a function of B, at
ρ = 0.6, A = 7, m = 10 and m= 0.1.

5 Conclusion

Based on analysis of known mechanisms for aggregation is proposed adaptive mechanism for aggregation with retransmission of fragments. The performance of this mechanism for aggregation is examined through simulations which have been developed by GPSS model.

The results confirm the correctness of the proposed approach in developing the A-AFR mechanism, i.e. aggregation to be done after reaching a certain threshold of utilization $\square^*$(respectively over certain intensity of packets arrival).

The presence of correlation between the time of feedback and the error burst length through transmission leads to non-trivial results, such as minimizing delays. This can be very important when designing new communication applications for operating in wireless networks using the proposed adaptive mechanism for aggregation.

5 Open Problem

Up comming wireless computer networks offer more high-speed data transmission in the physical layer (PHY) and using highly efficient protocols in the Medium Access Control layer (MAC) to access the communication medium.

High-speed of physical layer does not lead directly to increased efficiency of the MAC layer. The reason is that increasing speed leads to faster transmission of the MAC part in frame (user data), but the transmission time of PHY header and the back off time has not decreased substantially. For example, the new 802.11n standard offers speeds up to 600 Mbps. Transmission time of PHY header, however, is 48 µs. The maximum size of frame is limited to 7955 B. Thus, at a speed 150 Mbps, the time for transmitting user data is 424 µs, which means the proportion of transmission time for PHY header in frame is more than 10%. It is known that even under the best conditions, the efficiency of MAC layer (MAC_Layer_Speed/PHY_Layer_Speed) in 802.11n fall from 42% at a speed of 54Mbps to only 10% at speed of 432Mbps [9].

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