

Optimal MAC Packet Size in Wireless LAN

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Abstract - This paper presents an analytical method of optimal breaking of a Transmission Control Protocol (TCP)/Internet Protocol (IP) message into medium access control (MAC) packets in wireless networks (IEEE 802.11b wireless local area network standard), and an empirical, measurement-based, characterization of the average throughput as a function of the packet size.

TCP performs reasonably well over the wired networks where packet losses are mainly due to network congestion. However, TCP suffers significant throughput degradation over wireless networks where packet losses are also due to transmission errors.

In wireless LANs the MAC protocol is the main element that determines the efficiency in sharing the limited communication bandwidth of the wireless channel. Our analysis and simulations show that changing packet size could decrease/increase throughput and packet loss.

Wireless network links are characterized by rapidly time varying channel conditions, therefore, static link control techniques that make sense in well behaved wired links do not necessarily apply to wireless links. Adaptive sizing of the MAC layer frame in the presence of varying channel noise indeed has a large impact on the user seen throughput.

I. INTRODUCTION

In this paper, we focus our problem on performance of wireless local area networks (WLANs), which are becoming more and more popular. WLANs based on IEEE 802.11 standard are the most popular WLANs on the market. Reason for that can be found in prices, simplicity and matured technology, but the greatest advantage is they can be deployed in hot spots areas and offer performance comparable to wired local area networks (LANs), [8].

In the regular wireless local area network, some hosts may be far away from their access point so that the quality of their radio transmissions is low. There are some more reasons that cause data degradation, in our case, from the nominal 11 Mb/s rate to 5.5, 2 or 1 Mb/s. If there is at least one host with a lower rate, an 802.11 cell presents *performance anomaly*: the throughput of all hosts transmitting at the higher rate is degraded below the level of the lower rate, [7]. The reason for this anomaly is the basic CSMA/CA channel access method that guarantees that the long-term channel access probability is equal for all hosts.

One of the most common misconceptions about 802.11b is that the throughput is 11 Mb/s. The 11 Mb/s only refers

to the radio data rate of the packets on the physical layer. The throughput offered to the user of the IEEE 802.11b is significantly different. Maximum throughput that can be achieved in 11 Mbps (IEEE 802.11b), from user point of view, is about 6 Mbps. The efficiency of IEEE 802.11 WLAN is in sharp contrast to wired technologies where 10 Mbps Ethernet (802.3) link offers the users almost 10 Mb/s, [4].

This paper, presents an analytical and an empirical, measurement and simulation based characterization of the average throughput as a function of the packet size.

The rest of this paper is organized as follows. In section II we describe the IEEE 802.11 medium access methods. The DCF and PCF methods are described in order to provide reliable service to stations.

Throughput analysis is done in section III. Section IV describes simulation results. Some discussion about throughput improvement, through concatenation method, is analyzed in section V, so the paper is summarized in section VI.

II. 802.11 MEDIUM ACCESS METHODS

The 802.11 protocol (Media Access Control layer - MAC), is extremely robust and feature rich. It includes Sequence Control and Retry fields supporting a feature called MAC-layer acknowledge that minimizes interference and maximizes usage of the bandwidth available on the wireless channel.

The 802.11 standard ensures that all stations implements access methods for sharing the air medium. For low intensity traffic DCF is a better choice, but for real-time applications is better to use PCF.

A. Distributed Coordination Function (DCF)

The 802.11 standard makes its mandatory that all stations implement DCF, a form of Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). The reason for CA is because it's impossible to measure collisions on air medium. The main purpose is to avoid having stations transmit at the same time.

The DCF is quite more complex than this. An 802.11 station utilizes information it reaches from other frames that stations are sending over wireless link. In the control field of each frame, there is a duration field that a sending station places a value in, to indicate how long the station

will require the medium. As a part of making a decision on whether to transmit a frame, a station must see that the time associated with the duration value of the last frame sent has expired, as well as sense that no physical transmission taking place. The duration field enables stations to reserve medium for subsequent frames of some specific 802.11.

Because of this nature, DCF supports the transmission of asynchronous signals, [3].

B. Point Coordination Function (PCF)

As an optional access method, the 802.11 defines the PCF which enables the transmission of time-sensitive information. With PCF, a point coordinator within the access point controls which stations can transmit during any give period of time. Within a time period called the contention free period, the point coordinator may first poll station A, and during a specific period of time station A can transmit data frames (and no other station can send anything). The point coordinator will then poll the next station and continue down the polling list, while letting each station to have a chance to send data, [1].

Thus, PCF is a contention-free protocol and enables station to transmit data frames synchronously, with regular time delays between data frame transmissions. This makes it possible to more effectively support information flows, such as video and control mechanisms, having stiffer synchronization requirements, [3].

III. THROUGHPUT ANALYSIS

Let the TH indicates theoretical maximum throughput. TH could be defined as maximum number of LLC PDU (MSDU) that can be transmitted in a time unit. So we can write:

$$TH_{TL} = \frac{\beta}{\beta + \alpha} \times TH_{802.11b}, \left[\frac{\text{bit}}{\text{s}} \right],$$

where

- TH_{TL} is transport layer throughput;
- α is the total overhead above MAC layer;
- β is the transport layer datagram size, and
- $TH_{802.11b}$ is theoretical maximum throughput of 802.11b physical layer (we assume maximum throughput of physical layer to be the same as maximum throughput of MAC layer).

Also, some conditions for $TH_{802.11b}$ must be applied, [4]

- Bit error ratio (BER) is zero;
- There are no losses due to collisions;
- No packet loss occurs due to buffer overflow at the receiving node;
- Sending node always has sufficient packets to send;
- The MAC layer does not use fragmentation;
- Management frames such as beacon and association frames are not considered.

A. Calculation of $TH_{802.11b}$

To obtain the maximal throughput, we will divide the MSDU by the time it takes it to transmit it, [4]:

$$TH_{802.11b} = \frac{\text{MSDU}}{\text{Delay_per_MSDU}},$$

where from "Fig. 1":

$$\text{Delay_per_MSDU} = (T_{\text{DIFS}} + T_{\text{SIFS}} + T_{\text{BO}} + T_{\text{CTS}} + T_{\text{ACK}} + T_{\text{RTS}} + T_{\text{DATA}}) \times 10^{-6} [\text{s}]$$

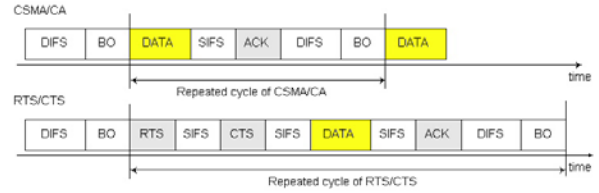


Fig. 1 Timing diagram for CSMA/CA and RTS/CTS

It can be simplified to a function of the MSDU size in bytes, x , as:

$$\text{Delay_per_MSDU}(x) = (ax + b) \times 10^{-6} [\text{s}],$$

where a and b are parameters with values given in table ("Table 2"), [4].

We can get theoretical maximum throughput by dividing the number of bits in MSDU by the total delay:

$$TH_{802.11b}(x) = \frac{8x}{ax + b} \times 10^6 \left[\frac{\text{bit}}{\text{s}} \right].$$

Table ("Table 1") presents throughput values for HR-DSSS CSMA/CA access method. Like is mentioned before i.e., when packet size tends to infinity throughput tends to 11 Mbps.

TABLE 1
ANALYTICAL RESULTS OF THROUGHPUT FOR IEEE 802.11b (CSMA/CA)

Throughput, Mbps	MAC Packet Size, byte
0,546	64
0,83	100
1,54	200
2,71	400
3,19	500
4,35	800
5,44	1200
6,06	1500

As well, when the MSDU tends to infinity, the $TH_{802.11b}$ is bounded by:

$$\lim_{x \rightarrow \infty} TH_{802.11b}(x) = \frac{8}{a} \times 10^6 \left[\frac{\text{bit}}{\text{s}} \right].$$

TABLE 2
 $TH_{802.11b}$ PARAMETERS FOR DIFFERENT MAC SCHEMES
AND SPREAD SPECTRUM TECHNOLOGIES

Scheme	Data Rate, Mbps	a	b
CSMA/CA			
FHSS	1	8,25	1179,5
	2	4,125	1039,5
DSSS	1	8	1138
	2	4	1002
HR-DSSS	5,5	1,45455	915,45
	11	0,72727	890,73
RTS/CTS			
FHSS	1	8,25	1763,5
	2	4,125	1623,25
DSSS	1	8	1814
	2	4	1678
HR-DSSS	5,5	1,45455	1591,45
	11	0,72727	1566,73

IV. SIMULATION RESULTS

This section describes network topology used for simulation ("Fig. 2"). Simulation was done in NS-2 simulator, [6]. We observed throughput as a function of MAC packet size. A simple simulation scheme has included infrastructure WLAN and IEEE 802.11b standards. We compared results for various MAC packet size. All the traffic flows in our simulation had the same priority.

The test bed consists of two nodes in wireless domain and one node in wired domain connected with the base station. These two nodes in wireless domain represent laptop computers, and wired node is server computer.

The idea was to continuously send data files from server to clients (nodes in wireless domain), changing MAC packet size, on the server side, for the purpose of monitoring transport layer throughput variety.

Clients only respond to server with acknowledgement, without sending any data.

Regardless of nodes mobility we kept them stationary, both at the different distance from the base station. We assumed that mobility of the nodes, in the base station cell area, does not affect transport layer throughput significantly. Presumption is correct because the main goal was to measure throughput depending on MAC packet size, and not on mobility.

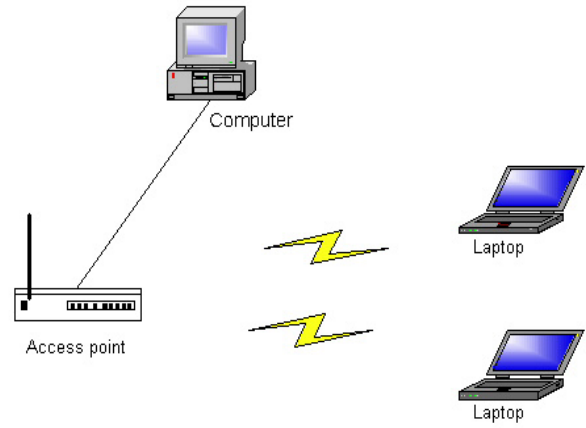


Fig. 2 Network topology

In an ideal situation throughput should rise proportionally with MAC Packet Size, but because of wireless instability and some other constraints it is not possible. Some of these constraints are:

- congestions;
- fading;
- multipath propagation of signal.

Measurements results (given in "Table 3") and their comparison with analytical results (given in "Table 1") are given in figure ("Fig. 3"). As seen the different MAC Packet Size gets throughputs according to their length. As the MAC Packet Size increases the throughput rises also.

TABLE 3
MEASUREMENT RESULTS

Throughput, Mbps	MAC Packet Size, byte
0,65	64
1,7	100
2,6	200
3,1	400
3,9	500
4,7	800
5,6	1200
5,8	1500

The throughput results, presented in this section, refer to ultimate throughput. It means it is equal to summary of throughputs of each node itself.

The difference between analytical and measurement based results can be explained with the fact that we took the peak value of throughput from the graph.

The main reason for difference in analytical and simulation results is BER (Bit Error Ratio). In other words we did not consider Bit Error Ratio during the calculation of throughput from trace file of NS-2.

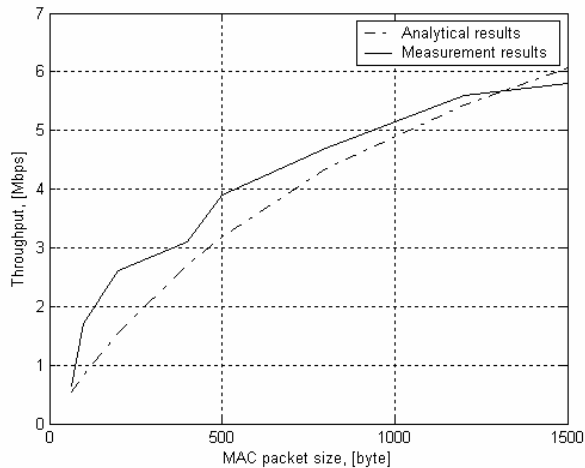


Fig. 3 Throughput of IEEE 802.11b versus MAC Packet Size

V. THROUGHPUT IMPROVEMENT USING CONCATENATION METHOD

The 802.11b specification also details frame format structure. This structure is designed to support DCF and PCF operation. The frame format consists of a MAC header and a frame body. The frame body contains the MAC service data unit (MSDU) from the higher layer protocols and has a maximum length of 2,048 bytes. The frame structure also sports a MAC header, which contains information on the frame type, destination addresses, and the length of the data payload.

Our intention is to discuss about concatenation data as a method for improving WLANs throughput.

Concatenation – the act of linking together – is the way to obtain required size of the data in the networking. In order to solve this problem many researchers use concatenation on different layers trying to solve the problem of small packets, [5]. These solutions are, through the layers:

- Link Layer
 - Packet Frame Grouping (PFG);
 - Packet Concatenation (PAC);
- IP Layer
 - Packet Concatenation on IP (PAC-IP);
- Transport Layer
 - Various algorithm and modifications.

A. PFG

The approach PFG was developed with the impact of improving the multimedia performance over WLANs. The key idea of PFG is to group small frames on the link level in order to share an overhead within the whole group.

The main advantages Packet Frame Grouping are: it does not increase latency of the packet delivery; does not limit to packets designed to the particular host.

B. PAC

The idea of PAC approach is to concatenate MAC layer frames into a superframe. The selection of the packet for

such concatenation is based on the same next hop address. Each packet includes MAC header and CRC in order to provide error independence for every packet.

VI. CONCLUSION AND FUTURE WORK

In this paper we presented calculation of throughput in IEEE 802.11b network depending on MAC packet size. We concluded that larger packet size provides greater transport layer throughput, although it is limited at about 6 Mbps for IEEE 802.11b. At the same time, MSDU is also limited, so there is no sense speaking of MSDU tending to infinity. Optimal MAC packet size mostly depends on environment and link state, but in most cases optimal MAC packet size would be 1500 bytes. Optimal refers to packet size for which throughput has maximum value.

The efficiency of most MAC protocols depends on traffic demands. There are no MAC protocols that can work well for different traffic flows. For traffic demands that are not intensive DCF access method is a better choice, but PCF is preferred when a large number of nodes is present and for real-time applications.

In further work we will be oriented for improving QoS in wireless LANs. QoS can be in some way implemented in IEEE 802.11b with still draft standard IEEE 802.11e. IEEE 802.11e provides two new coordination functions and allow better channel utilization with low jitter and reducing an end-to-end delay.

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