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APPLICATION OF THE SLOW START MECHANISM IN NOISY CHANNELS OF IEEE 802.11 WIRELESS COMPUTER NETWORKS

Purpose of the work. *The purpose of the work is to modify the slow start algorithm used in fiber-optic computer networks operated under the control of the TCP transport protocol in conditions of transmission without noise or in the presence of insignificant noise, to effectively apply the modified algorithm in wireless IEEE 802.11 networks in conditions of significant noise intensity.*

Methodology. *To reduce the overhead of transmitting information in IEEE 802.11 wireless networks, frame transmission is used in blocks. In wireline networks operating under the control of the TCP protocol, a mechanism that combines multiplicative and additive increases in the size of the congestion window is used to avoid packet loss. After an exponential increase in the number of packets, at some point the slow start threshold will be exceeded. Additional packets that enter the network get into congestion, and packet loss occurs. Therefore, after the threshold is exceeded, the window is halved and TCP switches to additive window growth. The general idea is that the TCP connection should work for as long as possible with a window size close to optimal one – not too small, so that the throughput is not low, and not too large so that congestion does not occur.*

Scientific novelty. *In the proposed article, we use a discrete-time Gaussian memoryless channel as a channel for transmitting frame blocks from the station to the access point of the infrastructure domain. In such a channel, bit errors are independent and equally distributed over the bits of information of the transmitted block. To ensure the effective operation of the slow start algorithm in wireless IEEE 802.11 networks under conditions of significant noise intensity, a principle is proposed for determining the threshold value depending on the number of a given size frames in the transmission blocks and the level of external noise, which is determined by the bit error rate in the blocks. The algorithm is provided with the property of adapting to the noise level, and the number of frames in the transmitted blocks is automatically regulated by the algorithm.*

Conclusions. *A modified dynamic algorithm for operation of the infrastructure domain of a wireless network using probabilistic characteristics of the block transmission process has been formalized. The domain throughput has been estimated depending on the noise intensity and the lengths of frames in blocks. The quantitative characteristics of the throughput reduction with increasing noise intensity and decreasing frame length in blocks is calculated. The number of frames in blocks is automatically regulated by the algorithm depending on the noise intensity.*

Key words: *IEEE 802.11 wireless network, noisy channels, frame blocks, slow start, throughput.*

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ЗАСТОСУВАННЯ МЕХАНІЗМУ ПОВІЛЬНОГО СТАРТУ В ЗАШУМЛЕНИХ КАНАЛАХ БЕЗДРОТОВИХ КОМП'ЮТЕРНИХ МЕРЕЖ IEEE 802.11

Мета роботи. *Метою роботи є модифікація алгоритму повільного старту, який застосовується в оптоволоконних комп'ютерних мережах, що працюють під управлінням транспортного протоколу TCP в умовах передачі без шуму або при наявності незначного шуму, з метою ефективного застосування модифікованого алгоритму в бездротових IEEE 802.11 мережах в умовах дії шуму значної інтенсивності.*

Методологія. *Для зменшення накладніх витрат при передачі інформації в бездротових мережах IEEE 802.11 використовується передача фреймів блоками. В оптоволоконних мережах, що працюють під керуванням протоколу TCP, для боротьби з втратами пакетів використовується механізм, який об'єднує мультиплікативне і лінійне збільшення розміру вікна перевантаження. Після експоненційного збільшення кількості пакетів на якомусь етапі відбувається перевищення порогу повільного старту. Додаткові пакети, які потрапляють у мережу, попадають в затори, виника втрата пакетів. Тому після перевищення порогу вікно зменшується вдвічі і TCP переключається на адитивне збільшення вікна. Загальна ідея полягає в тому, щоб TCP з'єднання максимально довго працювало з розміром вікна, близьким до оптимального – не маленьким, щоб пропускна здатність не була малою, і не дуже великим, щоб не виникало заторів.*

Наукова новизна. *В пропонуємій роботі в якості канала для передачі блоків фреймів від станції до точки доступу ми використовуємо дискретний у часі Гаусовий канал без пам'яті. В такому каналі бітові помилки є незалежними і однаково розподіленими по бітах інформації передаваємого блоку. Для забезпечення ефективної роботи алгоритму повільного старту в бездротових IEEE 802.11 мережах в умовах дії шуму значної інтенсивності запропоновано принцип визначення величини порогу в залежності від кількості фреймів заданого розміру в блоках передачі та рівня зовнішнього шуму, який визначається швидкістю бітових помилок в блоках. Забезпечена властивість алгоритму адаптуватися до рівня шуму, при цьому кількість фреймів в передаваємих блоках регулюється алгоритмом автоматично в залежності від інтенсивності шуму в середовищі передачі інформації.*

Висновки. *Формалізовано модифікований динамічний алгоритм роботи інфраструктурного домену бездротової мережі з використанням імовірнісних характеристик процесу передачі блоків. Здійснено оцінку пропускної здатності домену в залежності від інтенсивності шуму і довжини фреймів в блоках. Показано, що пропускна здатність зменшується при збільшенні інтенсивності шуму і зменшенні довжини фреймів в блоках. Кількість фреймів в блоках регулюється алгоритмом автоматично в залежності від інтенсивності шуму.*

Ключові слова: *бездротова мережа IEEE 802.11, зашумлені канали, блоки фреймів, повільний старт, пропускна здатність.*

Relevance. Over the past decade, the world has witnessed rapid development of wireless computer networks, which provide reliable communication, high spectral efficiency, and low latency (Giordani M., 2020; Chien T.V., 2022). At the same time, approximately half of the traffic is transmitted by the networks of IEEE 802.11 (Wi-Fi) family of standards (Dianne S., 2020, p. 1).

The speed and noisy immunity of modern computer networks largely depend on the solution to the problem of congestion. In several protocols, including TCP, the signal of link congestion is the loss of packets. Moreover, in wireless networks due to the presence of noise in the information transmission channels, this happens quite often. In most protocols, high throughput corresponds to minimal packet loss. In practice, this means that the packet loss rate for fast TCP links is low, on average not exceeding 1 %, and at a value of 10 % the link stops working. However, packet loss of the order of 10 % and higher is common for IEEE 802.11 wireless networks. If this is not considered, congestion control schemes based on the control loss parameter will significantly slow down the operation of wireless networks. The study of this problem at different noise levels in the information transmission environment is relevant.

Analysis of recent research and publications. In infrastructure domains of wireless computer networks, which are based on the use of DSF (distributed coordination function) function and the CSMA/CA collision avoidance mechanism, the station sends a frame when the transmission channel is freed, after waiting for the end of the DIFS (distributed inter-frame space) interval and the operation of the transmission slot selection mechanism (backoff mechanism). If this frame is received correctly, the access point (AP) sends back an ACK after short inter-frame space (SIFS). This interval is necessary for the physical layer (PHY) of the access point to transition from the receiving state to the transmission state (Tinnirello, 2010, p. 1056). In the event of collisions or interference in the channel when the AP cannot decode the corrupted frame, it does not send back the ACK frame to the sending station (STA).

To reduce the transmission overhead, a transmission mechanism using blocks of frames (Block ACK – BTA) was proposed (Kim, 2008; Li, 2005). In the BTA scheme, a block of frames intended for one recipient can be sent without confirming the correct reception of each frame by the AP. After sending the entire block, the sender generates and sends a Block ACK Request (BAR) frame to the AP requesting the frame numbers in the block that were successfully received. After that, the AP responds to the STA with a Block ACK (BA) frame. This mechanism is generally like the mechanism for transmitting data segments using sliding windows, which is used in the TCP transport protocol (Kurose, 2013, p. 233).

The efficiency of using the block transmission mechanism is associated with a significant reduction in overhead because DIFS and backoff time intervals are used only for the first frame in the block and only one ACK (BA) is sent by the AP to the sender after receiving the entire block.

In (Engelstad, 2006, p. 556) the authors analyze the throughput in the saturation mode of the BTA scheme for an infrastructure network with a channel without interference. In block transmission, the sender STA competes for access to the channel for the first frame of the block. If it wins the access competition, the transmission of the first frame begins, and after receiving the ACK confirmation for it and a short SIFS interval, the STA transmits the entire block of frames, which is accompanied by a BAR service frame. The transmission of the block ends with a receipt of the BA frame from the AP. This mode is called protected.

After receiving the BA frame confirming correct reception of the entire block, the sender begins a new phase of contention for access to the channel to transmit the next block.

At the same time, all other stations in the infrastructure domain wait for the end of the BA frame transmission, then wait for the DIFS interval and reset their backoff counters to a minimum to start a new phase of contention for access to the channel.

If there is interference in the data transmission medium, senders still send the entire block and the BAR frame. Upon receiving this, the AP sends back the BA frame with information about which frames in the block were corrupted. If the sender (STA) successfully receives the BA, then those frames that were correctly transmitted in the block must be removed from the transmission queue and a new block must be formed for the next round of transmission.

In (Banchs, 2006, p. 1749) the block transmission mechanism, mathematically analyzed for an ideal channel, is extended to the case of additive white Gaussian noise in the channel. In (Chen, 2006, p. 2226) the authors analyzed the throughput of the Block ASK scheme for the noisy channel.

After receiving the BA frame, the next data block can be formed only by distorted frames from the previous data block. Such a block is called a variable block size (VBS). Its size changes with each subsequent transmission stage in time. On the other hand, a data block can also be formed from distorted frames supplemented with new frames that need to be transmitted. In this case, the size of such a block is determined by the transmission conditions and is maintained constantly. Such blocks are called fixed block size (FBS) blocks (Lee, 2007, p. 408).

Analysis of the process of staged transmission of frame blocks under noise conditions showed that optimal choice of block size depends on the noise intensity value and differs when using VBS and FBS block formation mechanisms. However, the intensity of external noise is difficult to determine in such cases, and the suboptimal choice of block size and the mechanism for their formation during the transmission process significantly reduces the resulting network throughput.

Purpose of the study. The purpose of the work is to modify the slow start algorithm used in fiber-optic computer networks operated under the control of the TCP transport protocol in conditions of transmission without noise or in the presence of insignificant noise, to effectively apply the modified algorithm in wireless IEEE 802.11 networks in conditions of significant noise intensity.

Presentation of the main material. The modern congestion control scheme was implemented in TCP protocol largely thanks to the work (Jacobson, 1988, p. 314). He found that packet loss is a reliable signal of congestion, even though information about it may arrive with some delay. The solution proposed by Jacobson combines multiplicative and linear increases in the congestion window size. After establishing a connection, the sender transmits a small window of no more than four segments. In the case of successful transmission, the window size is doubled after a time interval equal to the round-trip delay. After an exponential increase in the number of packets 2^n, at some stage the slow start threshold SST will be exceeded. Additional packets that enter the network will get into congestion and packet loss will occur. In this case, after exceeding the threshold, the window is halved and TCP switches to an additive window increase. In this mode, the window is incremented by one segment after a time T equal to the round-trip delay, i.e. each increment occurs after a delivery confirmation is received. This receiving is not rapid. The general idea is to keep the TCP connection running for a long as possible with a window size close to the optimal one – not too small to cause low throughput, and not too large to cause congestion (Tenenbaum, 2011, p. 552).

In this work, we use a discrete-time memoryless Gaussian channel as a channel for transmitting blocks of frames from the station to the access point. In such a channel, bit errors are independent and equally distributed over the bits of information of the transmitted block. The probability P_C of a block successful transmission containing k frames by analogy with (Khandetskyi, 2022, p. 136) can be written in the following form:

$$P_S = (1 - P_C)(1 - P_B) = (1 - P_C)(1 - P_b)^{kL}, \quad (1)$$

where P_C is the probability of collision in an ideal channel, P_B is the probability of frame block distortion due to the interference during transmission, P_b is the probability of distortion of one bit of transmitted information in the block (*BER* – bit error rate), L – is the frame length in bits.

Let us fix the value of P_C at a certain level and determine the influence of the noise intensity, which is characterized by the BER value, on the probability of successful transmission of frame blocks when using the slow start mechanism. Let us denote this probability as $P_{BS} = P_S / (1 - P_C)$.

First, consider the case of low noise intensity, $BER = 10^{-7}$. We will assume the frame length to be close to the nominal $L_1 = 12\,000$ bits and the value of the slow start threshold SST $k_S = 21$ frames in the block. The fact is that at $k = 20$ the value of $k \cdot P_{BS} = 19{,}53$. This value is in the range

$$(k - 0{,}5) < k \cdot P_{BS} \le k \quad (2)$$

and thus, we can assume that block containing $k = 20$ frames are statistically successful transmitted on average over the channel to the access point. The corresponding value of $k \cdot P_{BS} = 20{,}477$, i.e. we will assume that one frame from this block will be damaged, the condition for successful transmission of such block is not met. Therefore, the value of SST threshold is $k_S = 21$. At this point, the initial section of the exponential increase in the number of frames in the block ends and the value of k_S according to the existing algorithm is halved.

Additive increase in the number of frames in the block starts with $k = 10$. In this section, blocks with the number of frames $k = 10, 11, 12, \ldots, 20$ are sent sequentially, each increase in k in the block occurs after receiving a delivery confirmation. The next block with $k = 21$ again exceeds the SST, and the threshold is halved and the transmission stage with additive increase in the number of frames in the block is repeated. The number of frames transmitted at this stage is calculated as the sum of n terms of arithmetic progression, so the throughput can be determined by the formula:

$$S = \frac{(b_1 + b_n) \cdot n \cdot L}{2(n + 3) \cdot T}, \quad (3)$$

where b_1 and b_n are the lengths of the first and the last blocks transmitted in the section of additive increase in the number of frames in the block (in our case $b_1 = 10$ and $b_n = 20$), n is the number of blocks in this section, T is the round-trip delay, i.e. the time interval from sending the block to receiving the confirmation. In this formula, it is assumed that the value of the retransmission timer for a damaged block is equal to $3T$ (Tanenbaum, 2011, p. 598).

The algorithm of the network operation in a formalized form is as follows. In the initial section of exponential increase, the number of frames in the transmitted block at each step of transmission increases as $k_i = 2^i$, де $i = 1, \ldots, m$ is the step number. After reaching the SST threshold, which occurs at stage m, the number of frames in the block is halved, i.e. $k_{m+1} = k_S / 2$. The value of SST is determined according to the expression:

$$k_s (1 - P_b)^{k_s \cdot L} \le (k_s - 0{,}5). \quad (4)$$

In this case, the even value corresponding to the smallest difference between the left and right parts in equality (4) is taken as k_S.

In the section from $(m + 1)$ to the q-th step, the number of frames in each transmitted block increases by one (the section of additive increase). That is $k_{m+1} + j = k_S / 2 + j$ ($j = 1, 2, \ldots, q$). After the number of frames in the block exceeds the threshold value SST, when inequality (5) is satisfied with the conditions given above for (4)

$$\left(\frac{k_S}{2} + q\right)(1 - P_b)^{k_{m+1+q} \cdot L} \le \left(\frac{k_S}{2} - 0{,}5\right), \quad (5)$$

then the block length is halved, i.e. $k_{m+2+q} = k_S / 2$ and the section of additive growth of the number of frames in the block is repeated. Table 1 shows the calculation results obtained in accordance with the presented algorithm for different noise intensity (BER) and different length L of frames in the block. The number of frames in blocks, the value of the slow start threshold SST, and the number of blocks in the areas of additive growth of block sizes are determined by the algorithm dynamically, adapting

Table 1

Relative values of network throughput

BER	S [kbit]/T		
	$L = 5$ kbit	$L = 8$ kbit	$L = 12$ kbit
10^{-7}	98,9	123,5	141,4
$2 \cdot 10^{-7}$	66	78	91,6
$4 \cdot 10^{-7}$	41,8	50,4	60
$6 \cdot 10^{-7}$	34,5	40	45
10^{-6}	25	30	30,8
$2 \cdot 10^{-6}$	15,6	16	18
$6 \cdot 10^{-6}$	7,5	8	7,2

to the noise intensity in the wireless information transmission environment.

To determine the absolute values of the throughput S, it is necessary to calculate the value of round-trip delay T, which, according to (Engelstad, 2006, p. 557), can be expressed as follows:

$$T = T_{DIFS} + T_{C\bar{W}} + T_{ACK} + T_{SIFS} + (k_{av} + 1)T_F + + T_{BAR} + T_{SIFS} + T_{BA} + 2\delta, \quad (6)$$

where $T_F = T_{PHYhdr} + \frac{L}{R} + T_{SIFS}$, T_{PHYhdr} denotes the duration of the physical layer frame header transmission, R is the transmission rate in bits per second; $T_{C\bar{W}} = \frac{\sigma \cdot (CW_0 - 1)}{2}$ denotes the average duration of the backoff process, CW_0 is the width of the contention window in slots, σ is the slot duration; δ is the signal propagation delay over the communication channel, k_{av} is the average block length (in frames) in the area of additive block length increase.

The following data were used to calculate the T values (Li, 2005; Chang, 2012), given for IEEE 802.11n, 802.11ac technologies: $T_{SIFS} = 16$ μs, $\sigma = 9$ μs, $T_{DIFS} = 34$ μs, $PHYhdr = 20$ μs, $CW_{min} = 16$, $2\delta = 0{,}7$ μs, $T_{BAR} = 21{,}8$ μs, $T_{BA} = 31$ μs, $L = 12$ kbit, 8 kbit and 5 kbit. The value of R is taken equal to 500 Mbps.

When the BER and the frame length L increases, the value of k_{av} in formula (6) decreases. Thus, for the sequence of BER values given in the first column table 1, for the $L = 5$ kb they are: 24, 17,12,10,8,5,3; for $L = 8$ kb: 19,13,9,8,6,4,2; for $L = 12$ kb: 15,11,8,6,5,3,2.

The corresponding dependences of the network throughput S on the BER level for different values of L are shown in Fig. 1.

As can be seen from graphs above, the throughput decreases with increasing noise intensity and decreasing frame length in bocks. The number of frames in blocks is not explicitly present in the dependencies, this number is adjusted automatically by the algorithm depending on the noise intensity in the information transmission medium.

Conclusions and prospects for further research.

1. The slow start algorithm has been modified. In its classical form, it is based on the segment congestion control mechanism using windows and is used in wired networks operating under the control of the TCP protocol, in conditions of information transmission without noise or in the presence of insignificant noise. The modification was made to apply this algorithm in wireless IEEE 802.11 networks in conditions of significant noise intensity.

2. The principle of determining the value of the slow start threshold depending on the number of a

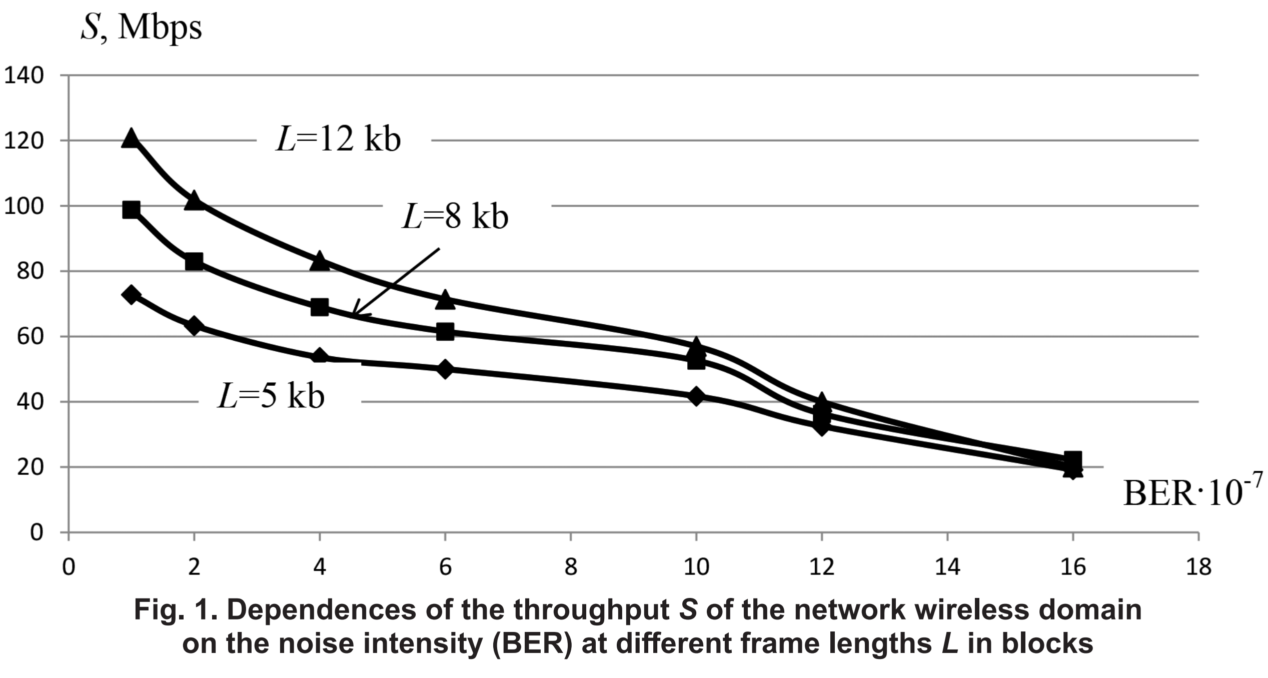

Fig. 1. Dependences of the throughput *S* of the network wireless domain on the noise intensity (BER) at different frame lengths *L* in blocks

given size frames in transmission blocks and the level of external noise which is reflected be the bit error rate (BER) in blocks, is proposed. The algorithm of the wireless network domain operation is formalized.

3. The domain throughput is estimated depending on the BER level and the length of frames in blocks. The algorithm has the property of adapting to the noise level, so the number of frames in blocks is automatically regulated by it.

4. The prospect of further research is to consider the influence of the collision intensity of frame blocks generated by different stations of the domain on the operation of the algorithm in order to determine the optimal conditions for transmitting frame blocks at significant levels of external noise.

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