Control of Exchange Messages in a Noisy Channel

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***Abstract*—Nowadays, almost all modern automation and communication systems use digital data channels and digital data transfer devices. Due to noise in the communication channel delays occur in data transmission or loss of information messages. This can lead to disastrous consequences, especially if the loss associated with the control messages. Therefore, evaluation of existing criteria on data reliability and methods of real data transmission speed to ensure safety and optimal conditions of trains on the stretch is important. Reducing the link layer protocols performance leads to a delay or loss of transmitted data. In turn, delays in data transmission channels in real-time systems can lead to dangerous consequences. The main reason for reducing the performance link layer protocols are the noise in communication channels. Interference can be both intentional (interference created by hackers), and unintentional. Currently, most networks carried over wireless communication channels. Another part there is a rapid development trend of mobile networks. As is known, wireless communication channels more vulnerable to interference than wired channels, concerning the question of studying the effect of noise on the performance of the link layer protocol is relevant, especially wireless links.**

I. Introduction

When transmitting digital messages for various reasons, there are with mistakes. If such reasons as imperfect digital data transmission systems, switching systems, poor quality of communication lines can be compensated for by their perfection, the presence of noise in communication channels - fundamentally irremovable factor.

Correctness of the data transfer via communication channels plays an important role. This is due to the existence of different standards and requirements for data transmission channels. The trend of development of transmission systems indicates a lack of performance evaluation methods link layer protocols based on the improvement of hardware and software components, in addition to these important methods is to develop recommendations on the link layer protocols settings (frame size, timeout, etc.).

Connecting between two workstations can be as simple as connecting any device with the Internet, the device can be in any size/configuration and operates on any operating system. But the main physical connections are through:

1) Optical fiber: is a method of transmitting information from one place to another by sending pulses of light, the main protocols that are used for this operation are SONET (Synchronous Optical Networking) & SDH (Synchronous Digital Hierarchy), they are standardized protocols that transfer multiple synchronized digital bit streams using lasers or LEDs. The two protocols are basicly the same, SONET is widely used in North America and SDH is used in the rest of the world. Due to the nature of these protocols and there transport-oriented features, they were the obvious choice for transporting the fixed length Asynchronous Transfer Mode (ATM) frames also known as cells. Optical Transport Network (OTN) is standardized by International Telecommunication Union (ITU) as G709, it was revised in 2009 so that it had considerable features and functions, among them is that G709 supports Forward Error Correction (FEC). These enhancements resulted from the need for greater packet client related features at various gigabits/s rates.

Fiber optic connections main advantages are:

- Extremely high throughput, proved reliable in transmitting data at rates that can reach 100 gigabits per second per channel.
- Very high resistance to noise, Because fiber does not conduct electrical current to transmit signals, it is unaffected by Electromagnetic interference. Thereby the BER(Bit Error Rate) is very low.
- Excellent security.
- Ability to carry signals for much longer distances before requiring repeaters. Depending on the type of fiber-optic cable used, segment lengths vary from 150 to 40,000 meters.

2) Copper via on the Ethernet: Ethernet introduced in 1980 and standardized in 1985, since then it has evolved to be the most widely used transport protocol for LANs, data center networks and carrier networks. Since then Ethernet speeds evolved from 10Mbps to 100 Gbps. Ethernet is a link layer protocol in the TCP/IP stack, describing how networked devices can format data for transmission to other network devices on the same network segment, and how to put that data out on the network connection. It touches both Layer 1 (the physical layer) and Layer 2 (the data link layer) on the OSI network protocol model. The major reason for the success of Ethernet in industry was the adoption of the Ethernet standard (IEEE 802.3), allowing for interoperability between different vendor's products (Carrier Sense Multiple Access with Collision Detection CSMA/CD, Access Method and Physical Layer Specifications) [15]. This specification allowed many different vendors to produce network interfaces and media that supported Ethernet.

Ethernet main characteristics are:

- Allows low-cost network implementations.
- Provides extensive topological flexibility for network installation.
- Guarantees successful interconnection and operation of standards-compliant products, regardless of manufacturer.

3) Wireless connection: A Wireless network is a wirefree connection between two or more devices that swap data via radio waves. It used to connect devices such as laptops, tablets and phones to the Internet, the business network and applications. Mobility is considered the primary benefit of wireless networks, mobile-based data communication is increasingly climbing in demand and has a rising share of overall network traffic. On the standardization side, 3GPP release and beyond have been approve by the ITU as ITU-Radio. The next generations will certainly have more enhancements and further increase in access data rates that will further enhance the usability of cloud access through mobile devices. The major technologies for mobile data access today are WiMAX, Bluetooth, ZigBee, LTE and Wi-Fi. As for WiFi Connections, they use the protocol 802.11 a,b,g,n and ac, these versions differ by bandwidth, frequency, stream data rate (speed) and distances that they cover [16]. However there still are many obstacles that stand on the way that make wireless connection not so efficient, mainly among them are noise and buildings which makes the BER high compared with other types of network connections.

II. PERFORMANCE OF COMMUNICATION PROTOCOLS

Performance of communication protocols is the speed of successful delivery of the payload from the message sender to the message receiver. Payload are the data which do not include service data, repeated frames (on revealing an error, the frames must be retransmitted), positive or negative acknowledgements on delivery or non-delivery of the frame to the message receiver, etc [11].

The bit error rate (BER) is the number of bit errors per unit time. The bit error ratio (also BER) is the number of bit errors divided by the total number of transferred bits during a studied time interval. BER is a unitless performance measure, often expressed as a percentage [9]. Real communication channels have level of noise in range within 10^{-2} <= BER <=10^{-7}. Depending on the quality of the communication channel, the real payload transmission rate can drastically decrease; this will, respectively, result in delays for the client. In this case, there is an urgent need to improve the real rate between the sender and receiver. The available methods for improving the performance of the communication level protocols were developed assuming that there is no impact of interference of different nature on data transmission channels [1] or the frame size variation [2], [3]. Quite important is the dependence of the frame size on the interference in the communication channels [8], data transmission modes [5], [6], [7].

For trusted data transmission, data transmission modes with feedback are commonly used, which means that the source of the message should validate that the message recipient has received the sent frame. If no confirmation is received after a certain period of time, it is considered that the frame has not been received correctly and should be sent again. Under these conditions, reliability of information transmission is expediently enhanced by retransmissions when it detects errors or loss of messages. Typically, flow control data transmission (Flow Control) is realized by introducing a mechanism of retransmissions at the link layer ISO/OSI model.

By increasing the frame size, you can increase the performance of the network, if suppose the events of distorted or lost frames are rarely [8]. Therefore, we can assume that for a given BER legitimately pose the problem of selection of the optimal frame size at which performance will approach the nominal data rate.

The protocols that use standby data transfer mode for the correctness of the delivery to the confirmation organize retransmissions of lost or corrupted frames. Correctness of the transmitted frame delivery for the message sender is a positive confirmation from the receiver of the message.

In organizing the retransmission of distorted frames sender of the message using the numbering of frames sent. Sender expects the receiver of the message for each frame a positive confirmation in the form of a service frame. Receiving a positive sender of acknowledgment message confirming that the frame received correctly. When sending a frame from the source of the message timer is starting, if not arrived at the expiration of a timer confirmation, then the frame is considering lost and sent again. The waiting time is determined by obtaining a confirmation timeout. In some protocols, the message receiver in the case of a frame with distorted data must send a negative confirmation, clearly indicating that the frame should be retransmitted (retransmission).

Organization of retransmissions is usually engaged system with automatic repeat request (ARQ - Automatic Repeat reQuest). There are several approaches to the process of the positive and negative exchange receipts: standby, returns to N steps and selective repeat. Return to steps N and selective repeat mechanism carried out in a "sliding window", in which the source sends a certain number of frames without waiting for the receipts; the number of frames determines the size of the window.

A simple method of transmission is the standby mode, which can be considered as a special case of the sliding window when the window size is 1. The source that sent the frame, waits for the ACK or NAK (ACK - Acknowledgment, NAK - Negative Acknowledgment) from the recipient and only then sends the next frame (or repeating frame that has been distorted).

The simplest case of such management is the use of a control channel protocol, running between two devices in a two-point connection. Composition of the studied parameters

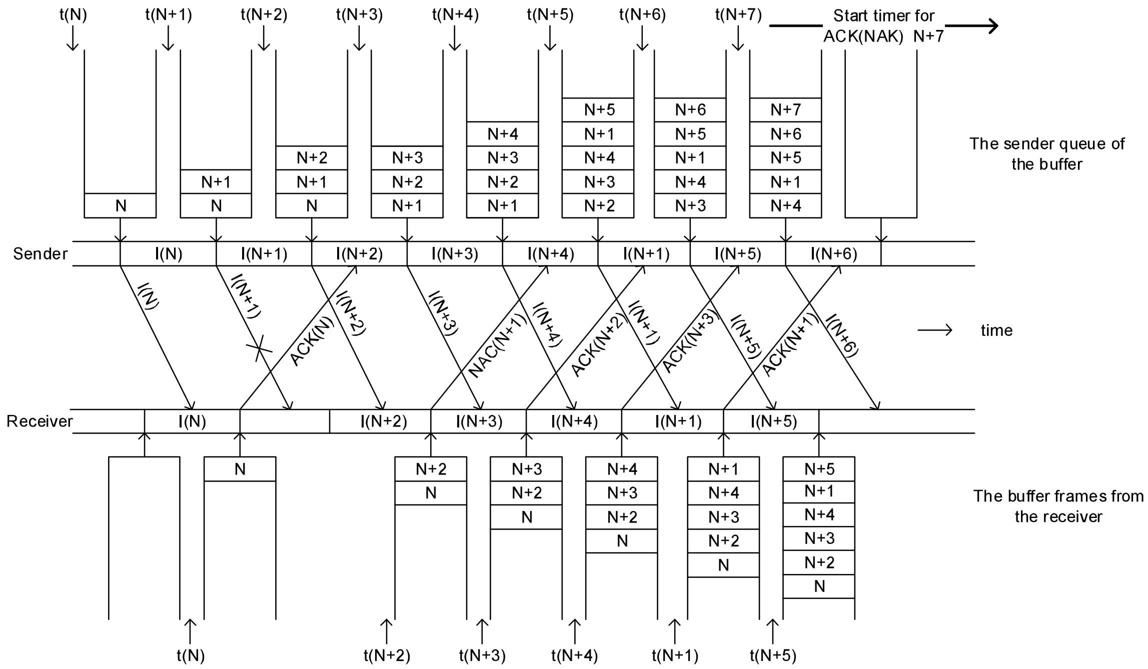

Fig. 1. Selective repeat method

includes a delay of data transmission between two devices, data transfer rate, probability and statistical traffic characteristics [1].

\For flow control used on the link layer, usually three methods: "Idle request", "go back N" and "selective repeat." The last two methods are the special cases of the sliding window technique [12].

III. SELECTIVE REPEAT

Consider two workstations connected communication channel, where the sender sends the message to the receiver. Each message is broken down into frames, which are sequentially forwarded to the receiver. The massage fragmentation linked with limitations associated the input buffer size of the receiver, as well as the brutal dependency of the transmission time on the length of messages under the presence of channel noise.

Consider the effect of certain events (error frame, an error in the ACK, duplication, etc.) On the performance of data-link protocols with respect to the method of selective repeat.

Selective repeat method (Fig. 1) is characterized by repeated sending only frames with errors or lost frames for which the message sender receives NAK or for which the time-out time has elapsed

In Fig. 1 shows an example of a sequence of frames, which illustrates the operation of the "selective repeat" method. Consider the sequence of works selective retry when an error occurs in the data frame:

1) Sender sends series frames and for each frame to the receiver sends a confirmation.
2) Let the frame N+1 is damaged or lost.
3) The receiver adopting frame N and frame N+2 detects that no frame N+1 and sends a NAK for frame N+1 sender.
4) Source received NAK for frame N+1 sends again frame N+1; while the source stops sending the next frame. In Fig. 1 upon receipt of NAK for frame N+1 is sent to frame N+4, and thereafter leave the frame N+1 instead of the frame N+5.
5) After this, set the timeout for the confirmation of the frame N+1.
6) If the time-out for the N+1 has expired and a receipt is not received or the source receives a NAK, then the frame N+1 re-sent to the receiver.
7) When receiving ACK for the frame N+1 power continues to send the next frame - N+5.

Selective repeat algorithm is more productive than one with a go back N, because this method minimizes the amount of retransmission frames. On the other hand, the receiver has to manage a buffer large enough to store all the frames received after sending them NAK frames as long as the frame is not distorted will be transmitted again. Receiver must also set itself received frames in the correct order.

IV. ANALYTICAL MODEL

Buffer memory for retention at the receiving side significantly affects the performance of the method. It is necessary not only to save the received frames, but also for sorting.

For example, if the window size is m * N (N - frame size, m - the number of frames in the window) bytes for sorting will need a minimum buffer size of m * N+log (m * N). Therefore there is a need definitions greatest probability of occurrence of errors in the window frame, for example, an error occurs in the first frame, or 10-th frame or in a frame in the window with size m.

The probability of errors in the k frames in the sequence of m frames is determined by the binomial distribution [2]:

$$P_{m,k} = C_k^m * p^m * q^{k-m}$$

where,

$$p = 1 - (1 - BER)^N \quad (1)$$

p – transmitting a frame without error probability over a communication channel with a predetermined BER, k – the number of error frames, $q = (1 - p)$ – probability of frame transmission with errors,

$C_m^k = \frac{m!}{(m-k)k!}$ – binomial coefficient.

We find the expected number of attempts until the first successful transmission of all m frames in the window [3]:

$$M(x) = \frac{1}{p} \quad (2)$$

The dispersion of the number of attempts before the first successful transmission of all m frames in the window:

$$D(x) = \frac{p}{(1-p)^2} \quad (3)$$

Determine the probability that at least one frame is distorted:

$$P_{1,m} = 1 - P_{0,m} \quad (4)$$

Table I shows the mean and variance for different BER at a rate of 1000 frames the window, the frame size 1120 bytes, calculated according to the formulas (2) and (3).

TABLE I. MEAN AND VARIANCE OF THE NUMBER OF ATTEMPTS BEFORE THE FIRST SUCCESSFUL TRANSMISSION OF ALL M FRAMES FOR DIFFERENT BER

№	BER	$M(x)$	$D(x)$
1	10^{-3}	673,902	219,757
2	10^{-4}	105,960	94,733
3	10^{-5}	11,137	11,013
4	10^{-6}	1,119	1,118
5	10^{-7}	0,111	0,111
6	10^{-8}	0,011	0,011
7	10^{-9}	0,001	0,001

For calculations according to the formulas (1 and 4) in finding the program was written Matlab environment, the likelihood of at least one of the frames in the transmission box is in error. Input parameters are defined as follows: frame size - 1120 bytes, the maximum size of the window - 1 000 frames, a step change in the window size - 5 frames. From Fig. 2, and on calculations made in the Matlab with environment, at high BER to use windowed mode communication channels is impractical, since the BER=10^{-4} window size is 6 frames; at BER = 10^{-5} window size is 23 frames; at BER = 10^{-6} window size is 57 frames. When BER<=10^{-8} probability of at least one error in the frame does not increase dramatically with increasing size of the window, and at BER = 10^{-9} value $P_{1,m}$ becomes more than 0,007.

Define the number of attempts to retransmit frames a given probability of successful transmission – u [1]:

$$P_{spec} = 1 - p^i,$$

where, P_{spec} – given the probability of the trusted of delivering frames, p^i – the probability in i step, thence,

$$u = \frac{\lg(1 - P_{spec})}{\lg p}$$

Fig. 3 for P_{spec} = 0.97 and BER = 10^{-3} with an increase in the frame size dramatically increases the number of repetitions. When BER <10^{-3} the number of repetitions of the frame does not exceed two. If the calculations to take BER=10^{-2}, the number of repetitions will increase dramatically, since the size of 500 bytes.

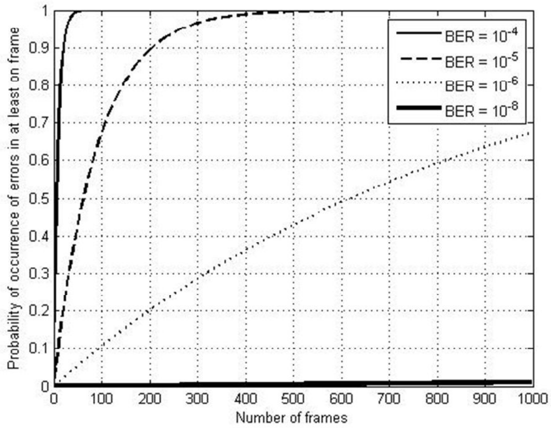

Fig. 2. The probability of at least one error in frames with selective repeat organization

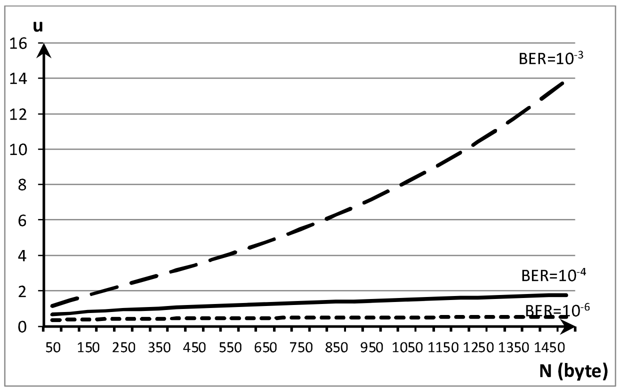

Fig. 3. The dependence of the number of transmissions required of successful transmission of the frame at different values of BER depending on the frame size

We consider a mathematical model of the method of selective repeat [14].

The probability of a successful transmission of a frame is determined similar to the method of sending the expectation by the formula (1) as in the case of an error in the frame is only sent back frame, for which the NAK.

The formula for calculating the actual data rate of the selective repeat the model is as follows:

$$V = \frac{(N-C)*m}{T_{sp}} \tag{5}$$

where N – frame length, C – the number of overhead bits in the frame, T_{sp} – the transmission of all the frames in a window with a selective repeat transmission method:

$$T_{sp} = \left(\frac{N}{R} + T_{int}\right) * \frac{m}{p} + \frac{S}{R_{\text{sig.prop}}} + T_{rc} \tag{6}$$

Timeout is an important component, the result of the choice of which affects the performance of data link protocols. Timeout value should not be small to avoid retransmissions, since the receipt will come later to the sender, and the source at the expiration of the timeout will re-send the same frame. However, this value should not be too great - it would lead to long periods of downtime in the communication channels in anticipation of the lost receipt.

In some standards [4] timeout is calculate:

$$T_{timeout} = RTT * B_{coof}$$

Here, RTT (Round-Trip Time) - the time of turnover is calculated as

$$RTT = 2 * t_{net},$$

where tnet is the time of data transmission with the maximum frame size permitted by the protocol standard.

Let us consider an example, when the source transmits a frame with the size of 1500 bytes by Ethernet DIX technology. For transmitting a frame of this size, 1,230.4 ms will be required. Substituting in equation for calculating the timeout $T_{timeout}$, let us assume that the weighting factor is 2; then Ttimeout = 2*1,230.4*2 = 4,921.6 ms. However, the receipt is sent back from the recipient to the message source (usually with the minimum frame size; in Ethernet DIX - 64 bytes), and the time for sending the receipt equals 67.2 ms. If account is taken of the fact that the delivery time from the sender to the recipient is equal to 1,230.4 ms and back - 67.2 ms, the gain in time will be 2,460.4 – 1,297.6 = 1,163.2 ms.

Suppose (Fig. 1), that after the departure of the last frame in the window (N+5) you must start the timer for making positive or NAK. In this case, you should use the following [3] ratio to calculate the RTT:

$$RTT = \frac{N}{R} + (m - \text{K}_{ACK}) * \left(\frac{N_{ACK}}{R} + 2 * T_{int}\right) + \frac{S}{R_{\text{sig.prop}}}$$

where K_{ACK} – number of the current ACK adopted by the sender, N_{rc} – current ACK adopted by the sender.

Loss ACK case is not as critical as in the method of return on N steps, as the loss or received in error is found when receiving the ACK following the ACK (N+2), (Fig. 1); at the where S – the length of the communication link, T_{int} – interframe gap, T_{rc} – time of waiting for the ACK of the last frame in the window, R – nominal speed link-layer protocol, $R_{sig.prop}$ – signal propagation speed on the transmission medium,

$$R_{\text{sig.prop}} = c * \mu,$$

where c - velocity of propagation in vacuum,
μ — the ratio of the actual speed of signal propagation in a vacuum [13].

In the method selective repeat need calculate timer for last frame in the window. Sender during the sending frames receives the ACK's (Fig. 1), in addition, after sending the frame starts a timer. If exposed timer was not receive ACK's sender will send again the same frame. sender of the frame saved in the frame buffer, which is not received ACK. Again, sent frame N+1 will be sent after frame N+3. The receiver of the message, taking the N+1 frame, detect existing or duplicate frames in the buffer and remove the newly adopted N+1 frame, thus sends an ACK(N+1) to sender. To detect duplicate frames receiver of the message should be stored in the buffer frames received all the frames for the current window, which correctly received from the sender of the message. In this regard, one can make a conclusion on the buffer size, namely the minimum size of the buffer should match the size of the window and to be equal to N * m.

In addition, a calculation using the formula (5) for Ethernet. They are illustrated in Fig. 4, which shows the actual link speed when using the method of selective repeat, taking into account the length of the transmission link. Input parameters are defined as follows: *C = 20 bytes, N = 1100 bytes, N_{rc} = 72 bytes, R = 10 Mb/s, $R_{sig.prog}$ = 1.98 * 10^8 m/sec.*

From the comparison in Fig. 4, we can conclude that by using the method of selective repeat the actual speed at BER> 10^{-5} close to nominal speed. These calculations determine the conditions for optimizing the real data transmission speed in the standard data link layer protocols, depending on the frame size and level of noise in the communication channels. These calculations can also serve as a recommendation to achieve the highest performance link-layer protocols.

V. Simulation Model

A. Formal model of experimental research

Using software Matlab Simulink library (communication tools) has developed a simulation model that implements selective repeat transmission mode.

Block "real speed protocol link layer", depending on the number of frames of repetitions determines the real speed to send the message using the formula:

$$V_{real} = \frac{m*N}{time} \tag{7}$$

where, V_{real} - the actual transfer rate from simulation model, *time* - time spent on transmitting the message, taken in the simulation.

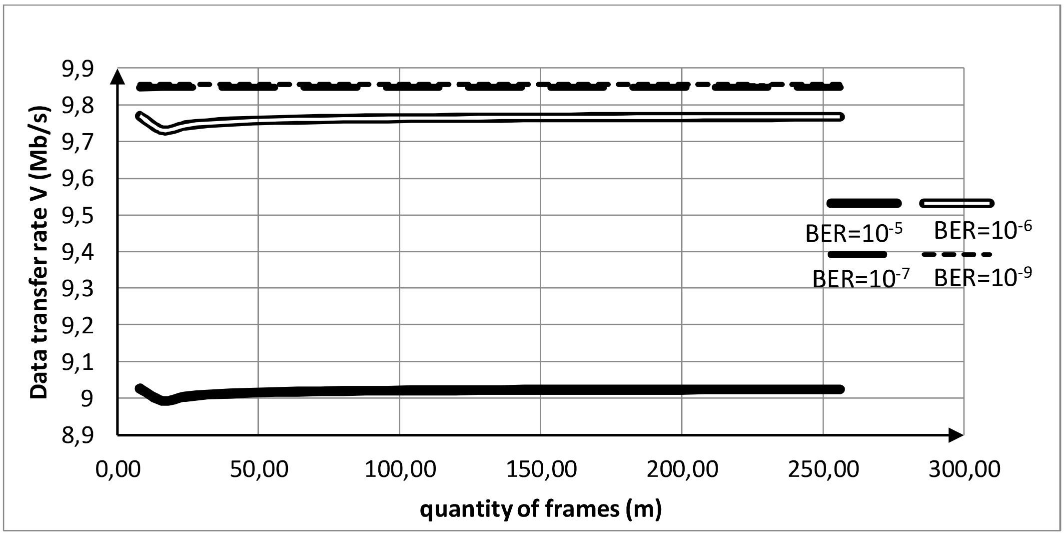

Fig. 4. Actual speed Ethernet communication channel for the method selective repeat

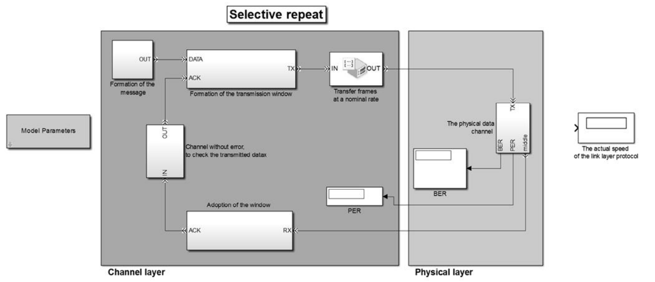

Fig. 5. The circuit model of "selective repeat"

The simulation model (Fig. 5) data for the method of "selective repeat" consists of the following units:

1) Formation of the message. In this unit, set the window size and frame size. Formed stream of frames with titles and numbers in the data queue.
2) Formation of the transmission window. In this block, the frames are sent depending on the receipt, if received an ACK, is sent to the next frame in the queue, the current frame is sent again if adopted NAK.
3) Transfer frames at a nominal rate. Unit transmits frames to the communication channel delays to achieve a given nominal speed.
4) Channel without error, to check the transmitted data. The unit checks if the frames in blocks of "acceptance window" and "the formation of a window for the transmission": if the frames are the same, the ACK is sent, otherwise - NAK.
5) Adoption of the window - the received frames in the window.
6) Model Parameters. In this block, set parameters to determine data transmission method.
7) PER (package error rate) - the probability of packet errors.
8) BER - bit error probability in the communication channel.
9) The physical data channel - channel interference, a given level of probability BER.
10) The actual speed of the link layer protocol.

The main purpose of "selective repeat" pattern is in imitation of the method at two lower OSI model levels, using the model in Simulink, including a subsystem "Data Link Layer" and "physical layer".

The model consists of the following units: generating the message, forming windows for transmission, transmission with the rated speed of frames without errors channel for checking the transmitted data, acceptance window, PER. The subsystem "channel level" posts forming unit generates a message and breaks them down into N frames. Each frame has a unique serial number that identifies it. User data is a constant equal to 1,120 bytes. In this model, the window size range, and BER.

forming a window unit for transmitting controls transmission and retransmission of data packets, and based on ACK- NAK messages.

This subsystem uses the principle of operation of the scanning condition thus controls transmission and retransmission of frames. Subsystem has the following states:

- Transfer - to set a new frame of data to be sent;
- Repeat - retransmission of the data frame in the communication channel;
- Waiting- expectation of error-free receipt of the communication channel after the successful transfer of the last frame in the window;
- Simple - successfully passed all of the frames in the window.

The principle of operation for a "for forming the transmission window" is as follows: when receiving NAK frame for the window generation unit N in turn adds to the frame number N, without stopping the transmission of the current frame by one. For example, while receiving NAK for frame N, suppose window forming unit sent frame number N + 4 - then the frame will be sent first N + 4, then the frame number N and then the frame number N + 5.

Subsystem "physical layer" implements the encoding techniques for transmission over a data communication channel with a predetermined interference. Subsystem transmits the modulated symbol through the channel, errors in transmitted block BER, PER errors in transmitted block.

Output Subsystem "error-free channel for checking transmitted data" becomes the input to block the formation of the window for data transmission. This block compares each input frame with each output frame, determining via CRC-32 frame in the presence of errors. The model assumes that the receiver detects any error types are also possible omission errors. In case of undetected errors in the ACK input receiver of the message sends a positive acknowledgment of acceptance of the frame with no errors, if found errors in the frame, it sends a NAK on the same input.

Block "link layer protocol actual speed" shows the number of transmitted megabits per second (calculated according to the formula 7).

B. Formation of error frames in the transmitted stream

For Standby was selected input data generated as a pseudo-random numbers. Thus 1,000 frames formed and these images are recorded in a file in binary form. There have been several experiments, the number of repetitions in which no more than five, the number of repetitions for each time the pseudo-random manner during the formation of the frame with errors:

1) Single errors in frames at different BER:
   - BER = 10^{-3}, in each frame of from 3 to 10 erroneous bits
   - BER = 10^{-5}, in every fifth frame has a bit error
   - BER = 10^{-7}, in only one frame of the 100 have bit error
2) Multiple errors in the frames at different BER:
   - BER = 10^{-3}, in every 10th frame has from about 10 to 30 erroneous bits
   - BER = 10^{-5}, in every 50th frame is from 5 to 10 erroneous bits
   - BER = 10^{-7}, in only one frame of the 100 have 10 bit error
3) Packet error frames at different BER:
   - BER = 10^{-3}, every 5th frame packet has errors
   - BER = 10^{-5}, every 50th frame has burst errors

Frame size was chosen 1120 bytes, window size is varied for different values of BER.

C. Interpretation of the measurement results

Simulation models for the three transmission modes have been designed based on real data link layer protocols. For the model of selective repeat the basis was taken HDLC protocol.

Output unit "real speed link protocol" (Fig. 6, 7) are collected in a file for comparison with the results of analytical modeling. The simulation model noise in communication channels were manually entered data to generate error frames in the stream are listed in section 4.B. The maximum number of retries on error equal to five has been found in the frames. For example, the first time the source of messages sent to the frame, the recipient of the message, an error is detected in the frame, sends a NAK message to the source, in turn, the source sends the frame again - can maximally be five such repetitions.

Fig. 6 shows a comparison of the results of analytical and simulation models at the level of BER = 10^{-5} for the selective repeat. At the same time it dispatched 3,000 personnel, including both the correct and incorrect frames. Window size ranging from 46 to 1500 with the increased pitch of the frame 2.

Because of the selective repeat the simulation results (Fig. 6), we see the results of most of the matches, but also there is a difference in speed. The difference in speed is associated with repeat - it is in these areas was less than the number of repetitions. Due to the high probability of errors the number of repeats ranged from 3 to 5 times.

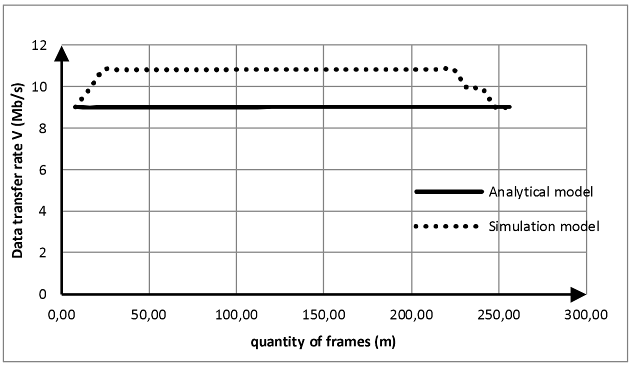

Fig. 6. Comparison of the results of mathematical and simulation models at the level of BER = 10^{-5}

The continuation of the comparison was to model the frame for other noise levels (in Fig. 7 to the value of BER = 10^{-7}). Jumping into the actual speed values are the result of a successful attempt at transmitting a frame 2 or 3, and said speed reduction to increase the number of attempts to 5. In general, the simulation results coincided with the results of the analytical modeling.

When the value of BER = 10^{-7}, (Fig. 7) noticeable surges in the actual speed of the simulation model, which is a result of uneven distribution of errors in the window, such as window size 100 to frame errors often met in the middle (in block 61) or the beginning of the window (in frame number 23). The analytical model was assumed that the errors are uniformly distributed in the transmission window.

At BER 10^{-9} (Fig. 8) actual useful data rate increases with the window size *(m)* which confirms the transmission in environments with such recommendation indicator using the maximum allowable window width standardized protocol link layer.

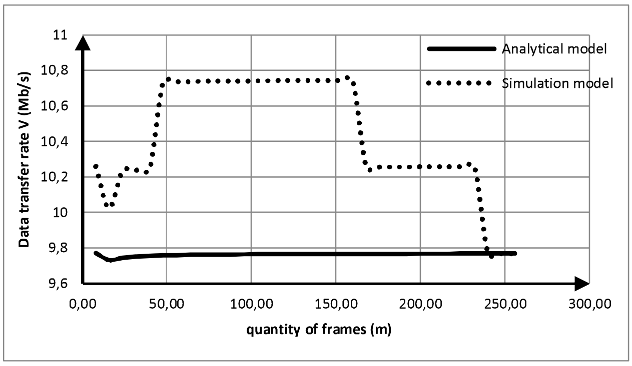

Fig. 7. Comparison of the results of mathematical and simulation models at the level of BER = 10^{-7}

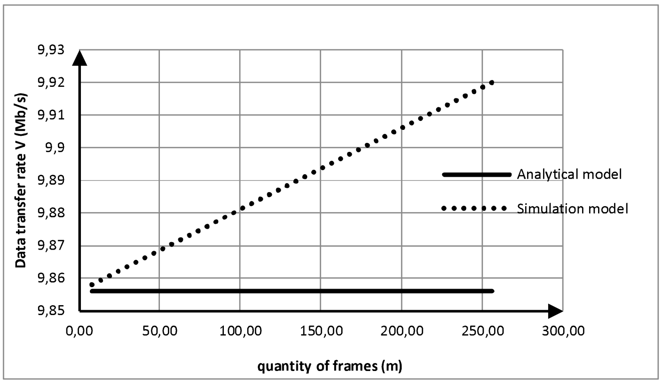

Fig. 8. Comparison of the results of mathematical and simulation models at the level of BER = 10^{-9}

Revealed the difference in the rates for the selective repeat, change the speed connected with errors. The difference in the actual speed of the two models is negligible, ranging from 0.1 to 1.2 Mb/s.

VI. CONCLUSION

In this paper gives the conditions for optimization of real data transmission speed in the standard data link layer protocols, depending on the frame size and level of noise in the communication channels. Recommendations for achieving the highest performance link-layer protocols for method of selective repeat.

Based on the comparisons made by the simulation and analytical models, we can conclude that the results of these methods are similar, although the methods of introducing errors into the communication channels in the two methods were different. The analytical model based on probabilistic interference estimates were evenly distributed, and in the simulation model error types (single, multiple, packet) to be entered manually, i.e. controlled by the user model. Having examined the influence in the simulations of different error types (single, multiple, burst errors) in the link layer protocols performance for the different methods of transmission have been identified as types of errors will affect the performance of data-link protocols.

Comparison of results simulation and analytical modeling, shows that their deviation does not exceed the boundaries of 11%, which confirms the consistency of the experiments.

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