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SIMULATION MODELING OF CONGESTION IN TELECOMMUNICATION SYSTEMS TO DETERMINE THE OPTIMAL PARAMETERS OF LINEAR NETWORK CODE

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Abstract

Simulink software environment is used as a tool for simulating congestion in networks that arise due to the presence of a “bottleneck” in the data transmission channel or the attacker's influence on the telecommunication system. The structure and functioning principle of the developed simulation model, as well as the data upload mechanism for their further use in determining the optimal parameters of a linear network code are described. A feature of the model is the implementation of the random early detection (RED) algorithm in it for queuing in its device, which simulates the “bottleneck”. Three cases were selected for modeling: load insufficient for serious losses, load close to maximum and load exceeding the capacity of the “bottleneck”. Based on the simulation modeling results, the average packet loss values are obtained and the data are downloaded in the form of text files. These data were transmitted as input parameters of the model, which determines the optimal parameters of the network code by the nature of packet losses in the downloaded text files. Linear network code parameters that are optimized include the length of the code sequence and the corrective ability of the code. The model estimates the probability of packet loss and calculates the code speed depending on the length of the code sequence and it’s correcting ability.

Keywords

Automatic retransmission request – Direct error correction – Queuing model

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Introduction

The purpose of this article is to study the effect of a linear network code on minimizing packet losses that occur due to the presence of the "bottleneck" in the data transmission channel or the attacker's influence on the telecommunication system[1]. The possibility of using a hybrid method for managing message delivery based on a linear network code is considered.

For such a study, it is necessary to assemble an isolated model that allows simulating the "bottleneck" in a transmission channel with moderate interference exposure with a minimum of external parameters. This model allows concluding that the parametric optimization of a linear network code is effective for ensuring reliable data transmission between subscribers of a telecommunication system in conditions of limited bandwidth and packet processing time. However, for a detailed study with a modification of all parameters, which are extremely numerous in this model, lengthy measurements will be required. At the same time, there is no guarantee that for the entire time the model has been functioning, the influence of many external factors has not changed and has not affected the experimental results. In this regard, the use of simulation is proposed, which will significantly accelerate the measurement process and will allow full control of the impact of various factors on the simulation model.

Structure and functioning principle of the developed simulation model

The model is designed on the basis of mathematically proven expressions from the queuing theory (QT) in the proven Simulink simulation modeling environment[2]. An analysis of the effectiveness of using Simulink to simulate real QT processes was considered.

The developed model is based on the queuing system M/M/1/K (one serving device with queue size K, while the arrival of packets is described by the Poisson process, and the processing time of the packets is subject to an exponential distribution). M/M/1 systems (Figure 1) are traditionally used to describe random natural processes [3 4], and a special case of such an M/M/1/K system (Figure 2) was used, which allows limiting the maximum size queue, which is required to describe a data link with the "bottleneck".

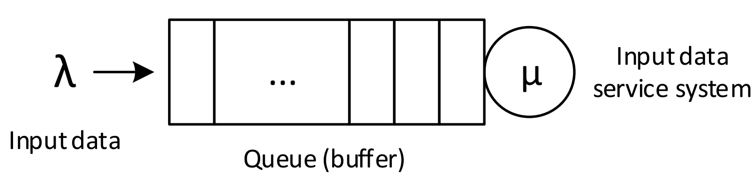

Figure 1
System M/M/1

[1] E. O. Karpukhin y N. V. Britvin, "Development of hybrid message delivery control method based on a linear network code", Telecommunication, Vol: 10 (2017): 30-36.
[2] S. Sameer, "Simulation: Analysis of Single Server Queuing Model", International Journal on Information Theory, Vol: 3 num 3 (2014): 47-54.
[3] M. Zukerman, Introduction to Queueing Theory and Stochastic Teletraffic Models. Hong Kong: City University of Hong Kong, EE Department. 2019.
[4] K. V. Ushanev y S. I. Makarenko, "Timeliness indicators of traffic service in the queuing system Pa/M/1 based on approximation of simulation results", Control, Communication and Security Systems, Vol: 1 (2016): 42-64.

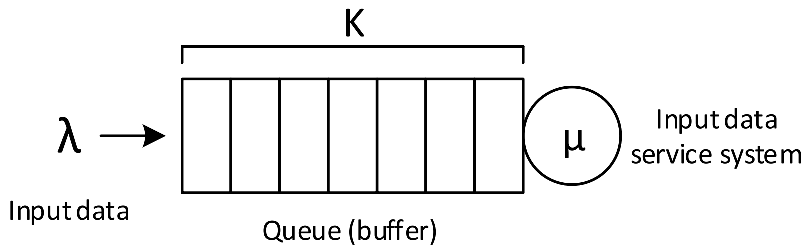

Figure 2
System M/M/1/K.

As the queue management algorithm, RED algorithm is selected. The most common algorithms are Tail Drop and RED, as well as RED analogues[5]. Tail Drop has several disadvantages that limit its use in modern data networks. They are ignoring the priorities placed in the packet headers, the phenomenon of TCP global synchronization and starvation[6]. In this regard, the RED algorithm was chosen, which will allow simulating the losses occurring in data transmission systems with the "bottleneck".

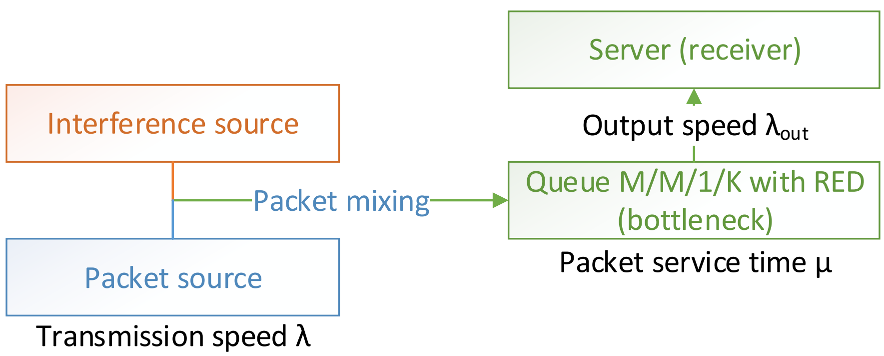

Figure 3
General scheme of M/M/1/K simulation model with the RED congestion control algorithm

The client (source of legitimate packets) creates packets with the interval specified at the model startup as an input parameter, while the intervals between the creations of each packet are not deterministic, since they obey the Poisson process.

In addition, packets from another source (interference source) are being sent to the queue simultaneously with traffic from the client. It can be legal processes that create a load on the network "bottleneck" and attacking influences (DDoS attacks). The distribution of such packets is also subordinate to the Poisson process with the ability to set the traffic arrival rate for this source. Thus, it becomes possible to simulate different degrees of load on the device and different degrees of client suppression. A detailed diagram of the sources of legitimate packets and interference is shown in the Figure 4.

[5] B. S. Goldstein; A. V. Pinchuk y A. L. Sukhovitsky, IP telephony. Saint Petersburg: BHV. 2014.
[6] Y. Gong, The quest for low-latency at both network edges: design, analysis, simulation and experiments. Paris: Department of Network and Computer Science, Telecom ParisTech. 2016.

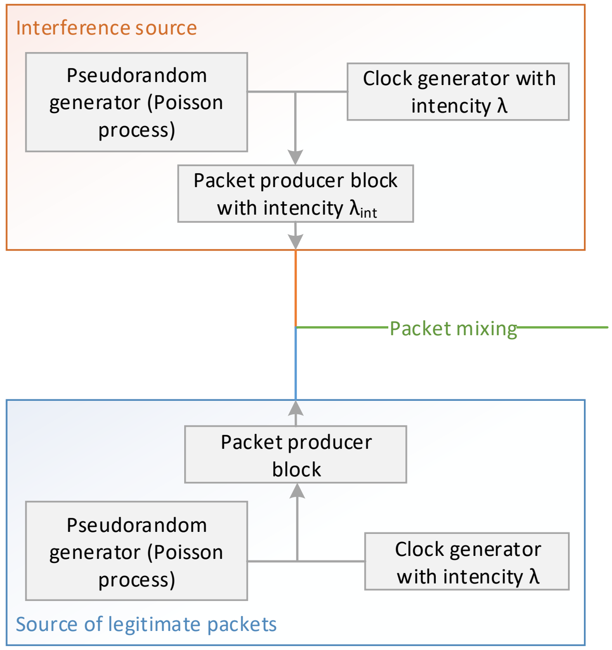

Figure 4
Detailed diagram of the source of legitimate packets and the source of interference.

It is important to note that this model allows obtaining the throughput of the queue and the intensity of data output from the system in isolation from the unit "bit". Instead of representing each packet as a set of bits of a certain volume, we can represent one packet as a kind of "entity", translating the speed of data service in the format of "packets per second". In the future, it will be enough to adjust only the speed of data service to obtain statistical results using the unit of measure “bit”.

The created packets arrive at the device with the queue, which is the “bottleneck” of the system, since it has a limited speed of processing traffic (limited bandwidth) (Figure 5). This device is based on the M/M/1/K system with the RED congestion control algorithm described above. Packets enter the queue and leave it according to the First In, First Out (FIFO) principle, that is, the packet that came first leaves the device. If the device’s bandwidth is less than the traffic intensity, packets begin to accumulate, which leads to an increase in delays. In the case of a limited queue length, packet accumulation leads to losses when the maximum load on the device is reached.

In this situation, the choice of the most effective congestion control algorithm plays a key role. Tail Drop drops packets when the queue is full. RED will mitigate loss peaks by dropping some packets before reaching the maximum queue capacity. Dropping starts from a certain limit set when the model starts.

Upon receipt of any packet in the queue, the main packet processing mechanism, RED algorithm, is adjusted. Possibility $p(x)$ of discarding the packet will be equal to[7]:

[7] D. Medhi y K. Ramasamy, Network Routing: Algorithms, Protocols, and Architectures. 2nd ed. Waltham: Morgan Kaufmann. 2017.

$$p(x) = \begin{cases} 0, \ 0 \leq x < t_{min} \\ \frac{x - t_{min}}{t_{max} - t_{min}} * p_{max}, \ t_{min} \leq x \leq t_{max}, \\ 1, \ t_{max} < x \end{cases} \quad (1)$$

where x is the queue congestion (in packets), t_{min} is the minimum number of packets (lower limit) in the queue, before the crossing of which discarding is not performed, t_{max} is the maximum number of packets (upper limit), after which all incoming traffic is discarded, p_{max} is the probability with which packets will be dropped if the queue is full between t_{min} and t_{max}.

At the time of discarding part of the traffic, one of the main output indicators of the system is being calculated (average probability of packet loss). For this indicator, the following formula is used:

$$P_l = \frac{n_c}{m - m_p}, \quad (2)$$

where P_l is the average probability of packet loss, n_c is the number of packets sent by the client, m is the total number of discarded (lost) packets, m_p is the number of dropped packets from the source of interference.

In the simulation model, such a solution is implemented as a separate RED unit, which receives the number of N packets in the queue at the moment. It also contains other necessary information about the model, for example, about the size of the maximum queue length K. At the output, the RED block decides whether to discard or receive the following packet: “0” if the packet is skipped, and “1” if it is discarded. This data goes to the mechanism of collecting a numerical array (or dump) and to the switch, which sends packets either to the queue or to the dump (in this way, feedback is implemented).

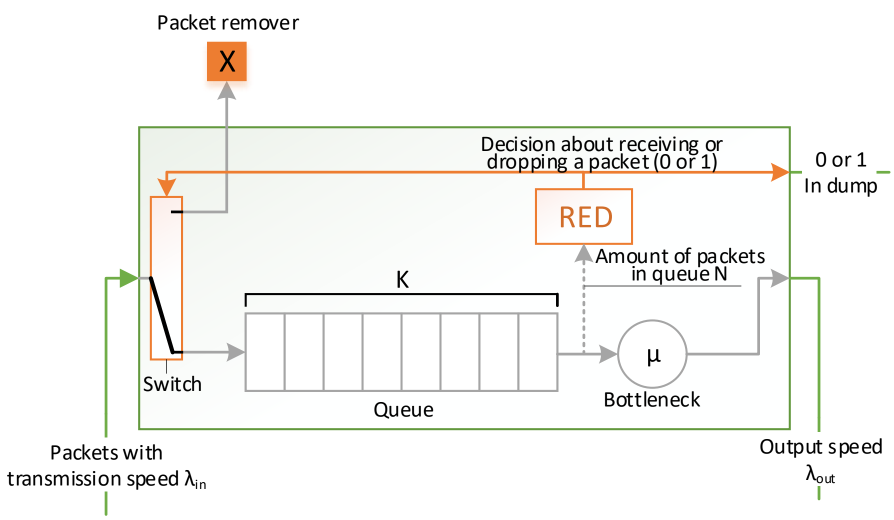

Figure 5
M/M/1/K queue with RED congestion control algorithm in simulation model.

When a packet leaves the queue, another output indicator is calculated: waiting time of the packet in the queue t_w, that when combined with delays introduced during transmission over the channel (t_d), allows calculating the total delay t:

$$t = t_w + t_d. \quad (3)$$

When a packet is dropped, and when it has successfully passed the queue, information about its state is written to the numerical array: 0 - when the packet was successfully passed, 1 - when it is lost (dropped). The experiment ends when a certain number of packets specified at the beginning of the simulation was sent from the client and each of them was either delivered or dropped. At the end of the experiment, the previously saved array is written in the form of a dump file (as an example, a fragment of such a dump is shown in the Figure 6). It will be used as input for a model that determines the optimal parameters of a linear network code for a specific combination of data transmission intensity, delays, losses, etc.

1;1;1;1;0;0;0;0;0;0;0;1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;1;1;0;0;0;1;1;1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
1;0;0;0;0;0;0;1;1;1;1;0;0;0;0;0;0;0;0;1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;1;0;0;1;0;1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;1;1;1;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;1;0;0;1;1;1;1;1;1;0;0;0;0;0;0;0;0;0;0;0;0;1;0;0;0;0;
1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;1;1;0;0;0;1;1;1;
1;0;1;1;0;0;0;0;0;0;0;0;0;1;0;0;0;0;0;1;0;1;1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;1;1;1;1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;1;0;0;0;0;1;0;0;0;0;1;
0;0;1;1;1;1;1;0;0;0;0;0;1;0;0;0;0;1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;1;
0;0;0;0;0;0;1;0;0;0;1;0;0;0;0;1;0;0;1;0;0;0;1;0;0;0;0;0;0;0;1;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;1;0;0;1;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;
0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;0;

Figure 6
Excerpt from a dump resulting from simulation modeling.

When the packet leaves the queue, the last two indicators are calculated: "server utilization" and packet processing rate (or output speed) $\lambda_{out} = \frac{1}{\mu}$. The workload of the queuing device ρ is calculated by the formula $\rho = \frac{\lambda}{\mu}$, (for systems with one queuing device), where λ is the data acquisition rate, μ is the maintenance time (how long the queuing device processes one packets). This parameter allows considering how busy the queuing device is within a dimensionless quantity. The presence of such a parameter allows making several important conclusions[8]:

1.- The intensity of the data processed by the system increases from until it reaches a maximum, and then quickly begins to decline (at this moment the queue is full and the probability of packet loss becomes extremely high).
2.- The delay starts to increase greatly when ρ approaching 1.
3.- The safest and most effective value ρ, which is worth trying to comply with, is about 0.5.

In addition, by measuring the number of packets leaving the system per second, we can get an "output" data transfer rate, which shows how much the queuing device slows down the flow ("bottleneck"). This characteristic will allow building dependencies of code parameters using a model to determine the optimal parameters of a linear network code. When plotting the graphs, the previously obtained probabilities of packet rejection (loss), delay and also the collected dump file will be considered.

[8] S. Mneimneh, Computer Networks M/M/1 with finite queue, Computer Science. New York: Hunter College of CUNY. 2008.

Simulation model results

To achieve the goal set at the beginning of the article, it is necessary to select the model parameters that will allow evaluating the effectiveness of the linear network code under different conditions of the environment in which the data is transmitted.

It is proposed to consider three states of the system: load on the channel is insufficient to cause serious losses; load on the channel is close to the maximum throughput of the "bottleneck" and load on the channel exceeds the maximum throughput of the "bottleneck". Serious losses are understood as losses of about 1% with a short-term sharp increase in the intensity of packet transmission in User Datagram Protocol (UDP)[9].

For the first state, the total load $\rho = 0.9$ (90 %) was selected, for the second it was $\rho = 0.99$ (99 %) and for the third it was $\rho = 1.1$ (110 %). Table 1 shows the initial parameters for the three states of the system described above.

Parameters	**ρ = 0.9**	**ρ = 0.99**	**ρ = 1.1**
Queue length	40		
Upper bound of RED	40		
Lower bound of RED	37		
P_{max} for RED	0.1		
Average service time of one packet, ms	1		
Legitimate client data transfer rate	0.5		
Intensity of data transmission by a source of interference	0.4	0.49	0,6

Table 1
Simulation model parameters

Lower and upper bounds of RED, as well as Pmax were selected according[10].

The intensity of data transmission in each of the states changed at the source of interference, as this allows one to more accurately determine changes in the nature of packet loss at the client with the most similar set of source data. Thus, it turns out to fix most of the parameters and change only the most significant ones, which increases the accuracy of the measurement.

Based on these initial parameters, a simulation of the process of transmitting data through the "bottleneck" was carried out. The results of the model are collected in three dumps for each of the states of the system and their analysis is carried out. The result of the analysis is shown in the Table 2.

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[9] B. S. Goldstein; M. A. Marshak; E. D. Mishin; N. A. Sokolov y A. V. Tum, "Performance indicators of a public multi-service communication network", Communication Technology, Vol: 3 num 4 (2009): 25-31 y V. Markovski; F. Xue y L. Trajkovic, "Simulation and analysis of packet loss in User Datagram Protocol transfers", The journal of supercomputing, Vol: 20 num 2 (2001): 175-196.

[10] J. M. Amigó; G. Duran; A. Giménez; O. Martínez-Bonastre y J. Valero, "Generalized TCP-RED dynamical model for Internet congestion control". Communications in Nonlinear Science and Numerical Simulation, Vol: 82 (2020): 1-19.

Analyzed parameter	ρ = 0.9	ρ = 0.99	ρ = 1.1
Packets sent	10003261	10003261	10003261
Lost client packets	13277	194415	927240
Lost client packets (%)	0.133%	1.944%	9.269%

Table 2
Results of the analysis of dumps obtained as a result of simulation model

As we can see from the results of simulation modeling, with an increase in ρ there is an increase in the number of lost packets, which becomes significantly higher when ρ > 1. In general, the model correctly implements the mass service process, since when the device is loaded ρ = 1.1, packet loss is about 9%. If we assume that 11 packets were sent, and received by the receiving side 10 (which corresponds to ρ = 1.1), we get the same 9% loss. The limit on sending packets of 10 million is associated with the duration of the modeling, which takes about an hour, but at the same time provides a sufficient sample size to search for optimal parameters of the linear network code.

Discussion of the search results for optimal parameters of the linear network code

The numerical results obtained after simulation modeling, as well as dumps of the data transfer process in the format of text files, are transferred to the model. According to the results of its work, the optimal parameters of the linear network code, such as the length and information sequence, are determined for each specific state of the system. The search for the optimal lengths of information (K) and code (N) sequences of a linear network code is carried out according to the following criteria: maximizing the speed of the code and minimizing the length of its sequence provided that the packet loss probability is zero. If $P_l = 0$, for a given pair of values (N, K), the code rate R is calculated and stored in a variable R_{max}. When the condition is repeated, a comparison is made with R and R_{max} with saving the value (N,K) only for that pair where the code speed is maximum. However, to reduce the data collection time, it is necessary to introduce another range of admissible values of R, which differ slightly from R_{max}. To do this, another variable is introduced into the algorithm for finding optimal values R_e. It allows choosing the pair (N, K) that will provide the minimum data collection time, which, in turn, depends on the length of the code sequence N.

Based on the results of the model's work, graphs of the dependence of the probability of packet loss on the length and corrective ability of the code are obtained. The optimal parameters of the linear network code are determined at which the probability of packet loss is zero. The dependence of the code speed on the length of the code sequence and the correcting ability of the code is found. The optimal points (N, K) of the linear network code were also determined for each of the three cases in which the code speed is close to the maximum or maximum. The size of the code sequence of packets in the model was limited to 100 packets, however, this parameter, as well as some others (corrective ability of the code, congestion of the queuing device), can be changed over a wide range of values.

Results for ρ = 0.9.

A graph of the dependence of the probability of packet loss on the length and the correcting ability of the code is obtained (Figure 7). A graph of the parameters of the linear network code was constructed for which the probability of packet loss is zero (Figure 8), and a graph of the dependence of the code speed on the length of its sequence and its corrective ability (Figure 9). The values of the optimal points for the linear network code are presented in the Table 3, where T is the corrective ability of the code, and R is the code speed. Packet drop probability means the impossibility of restoring a lost packet, which leads either to a second request or to the continuation of the codec without this packet.

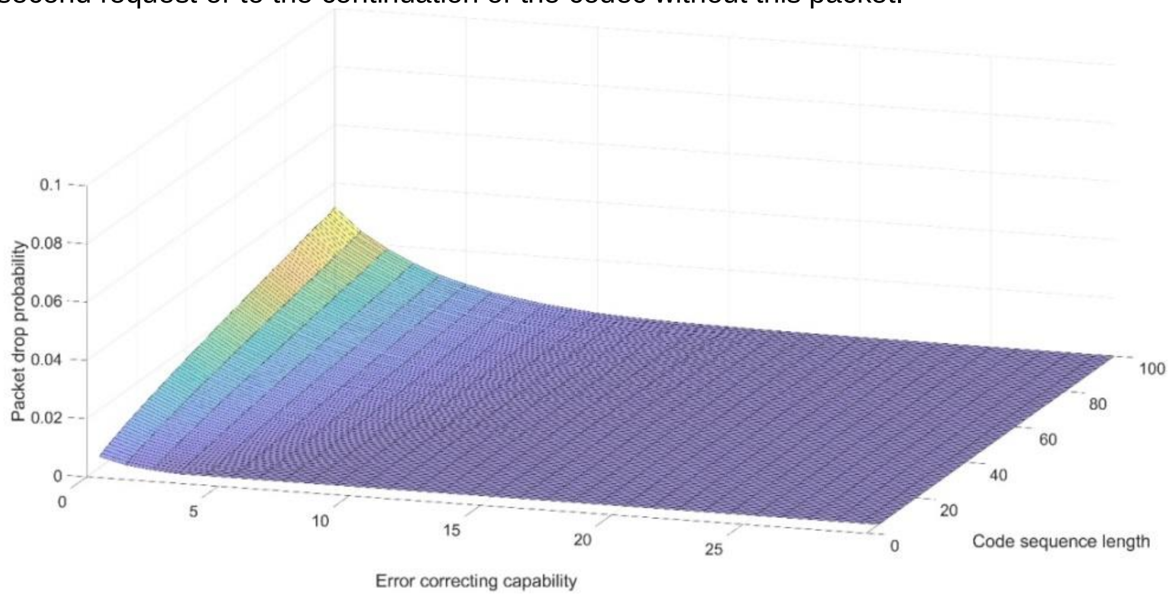

Figure 7
Graph of dependence of the probability of packet loss on the length and the correcting ability of the code for ρ = 0.9.

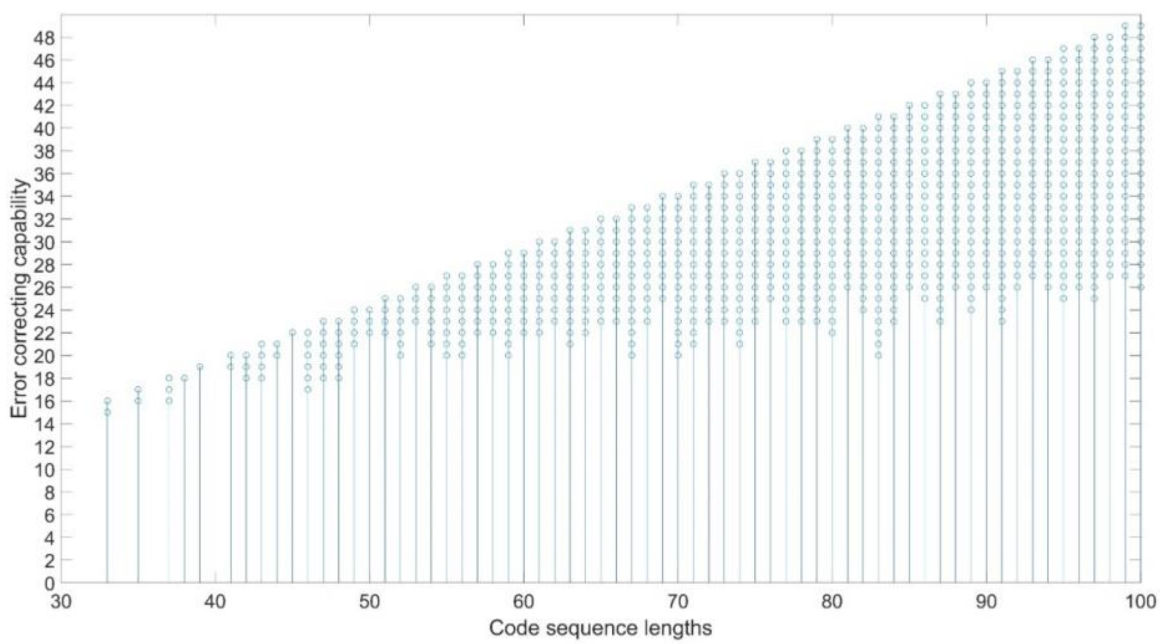

Figure 8
Set of optimal linear network code points for ρ = 0.9.

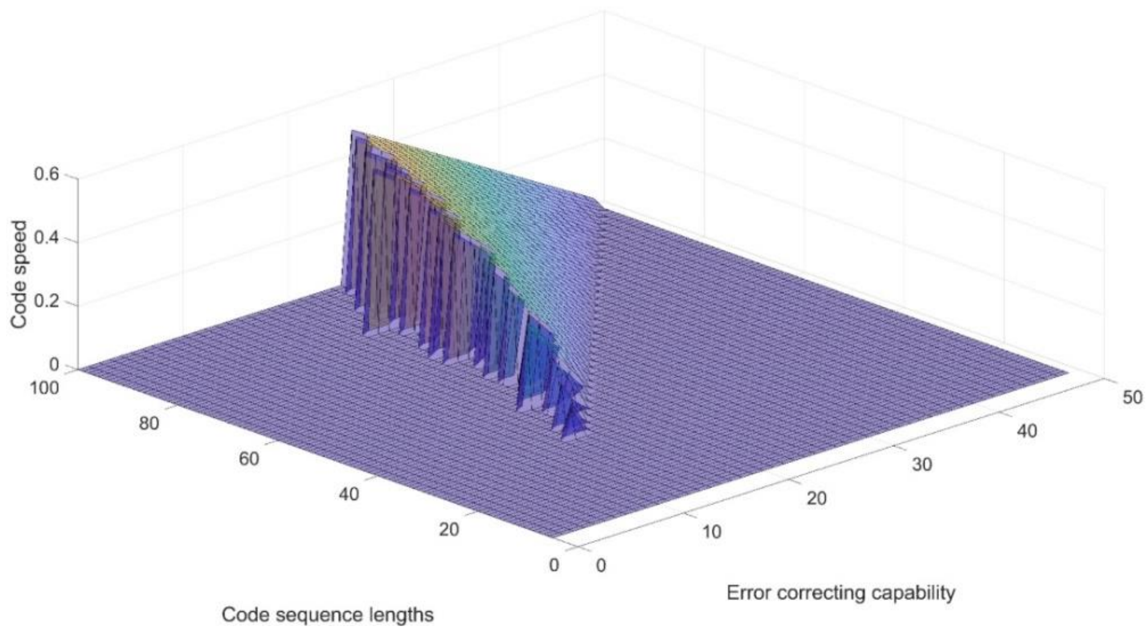

Figure 9
Graph of code speed versus code sequence length and code correcting ability
ρ = 0.9.

N	**T**	**R**
83	20	0.518
91	23	0.495
83	21	0.494
97	25	0.485
100	26	0.48

Table 3
Optimal points for ρ=0.9.

Due to the small amount of data transmission loss, a large correcting ability of the code is not required. There is a wide selection of different combinations of code sequence lengths and code corrective capabilities (Figure 8). When transmitting multimedia data (audio and video), a small percentage of losses allows abandoning the encoding of data at the recipient and significantly reduce overhead.

Results for ρ = 0.99

The graph of the dependence of the probability of packet loss on the length and corrective ability of the code is presented in the Figure 10. A graph of the parameters of a linear network code at which the probability of packet loss is zero is shown in the Figure 11. A graph of the dependence of the code speed on the length of the code sequence and the correcting ability of the code is in the Figure 12. The optimal parameters of the linear network code are shown in the Table. 4.

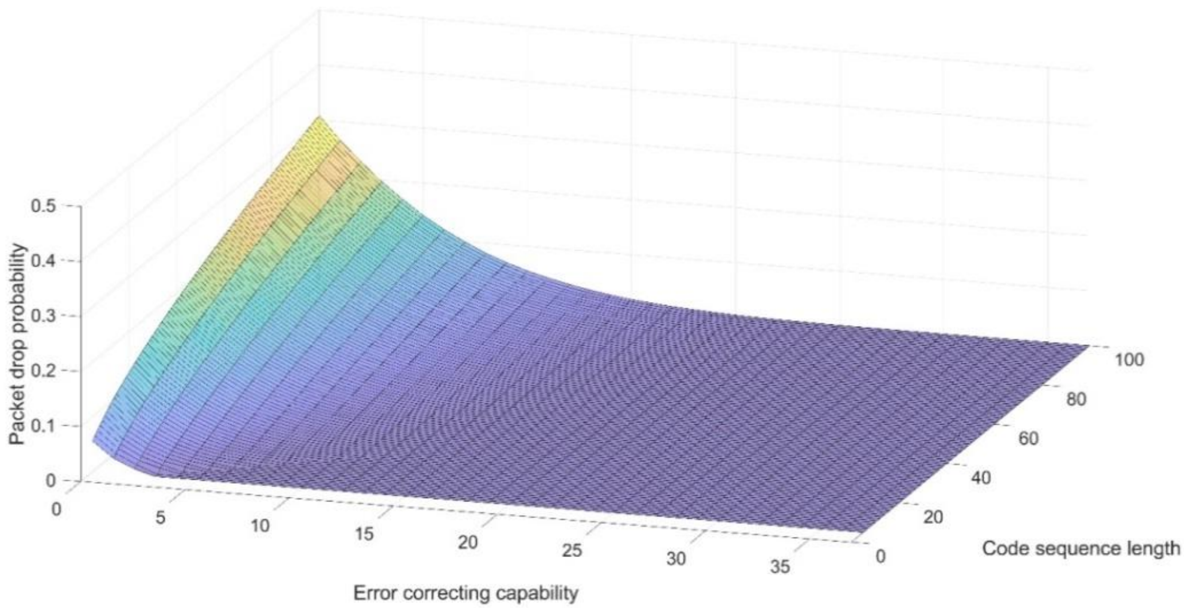

Figure 10
Graph of code speed versus code sequence length and code correcting ability

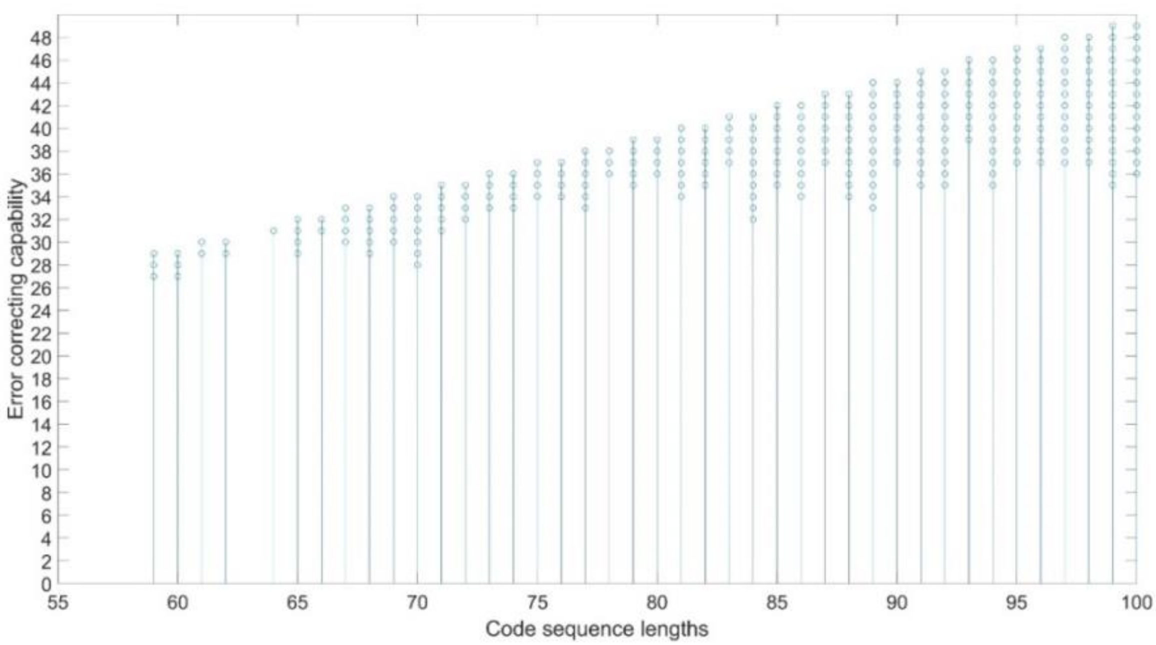

Figure 11
Set of optimal linear network code points for ρ = 0.99.

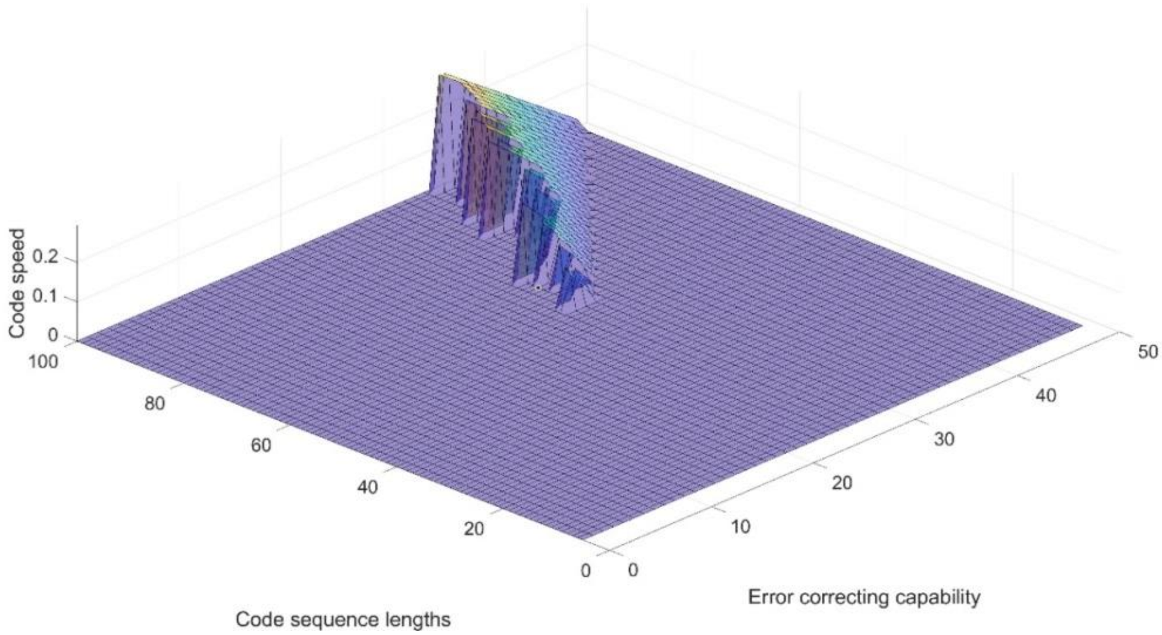

Figure 12
Graph of code speed versus code sequence length
and code correcting ability
ρ = 0.99.

N	T	R
99	35	0.293
100	36	0.28
99	36	0.273
100	37	0.26
89	33	0.258

Table 4
Optimal points for ρ = 0.99.

For ρ = 0.99 a moderate number of losses is characteristic, which becomes high when using a code with too little corrective ability, which requires code parameters like (100.28) or (89.23).

Results for ρ=1.1

The graph of the dependence of the probability of packet loss on the length and corrective ability of the code is presented in the Figure 13. A graph of the parameters of a linear network code at which the probability of packet loss is zero and a graph of the dependence of the code speed on the length of the code sequence and the correcting ability of the code are presented in the Figures 14 and 15. The optimal parameters of the linear network code are presented in the Table 5.

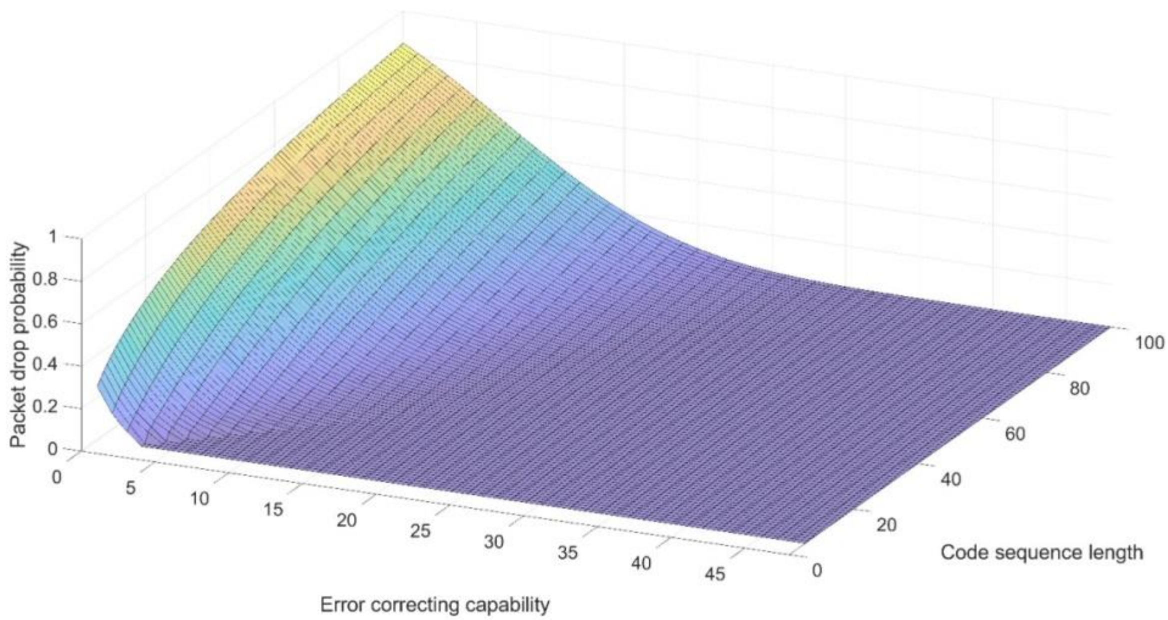

Figure 13
Graph of the probability of packet loss on the length and corrective ability of the code for ρ = 1.1.

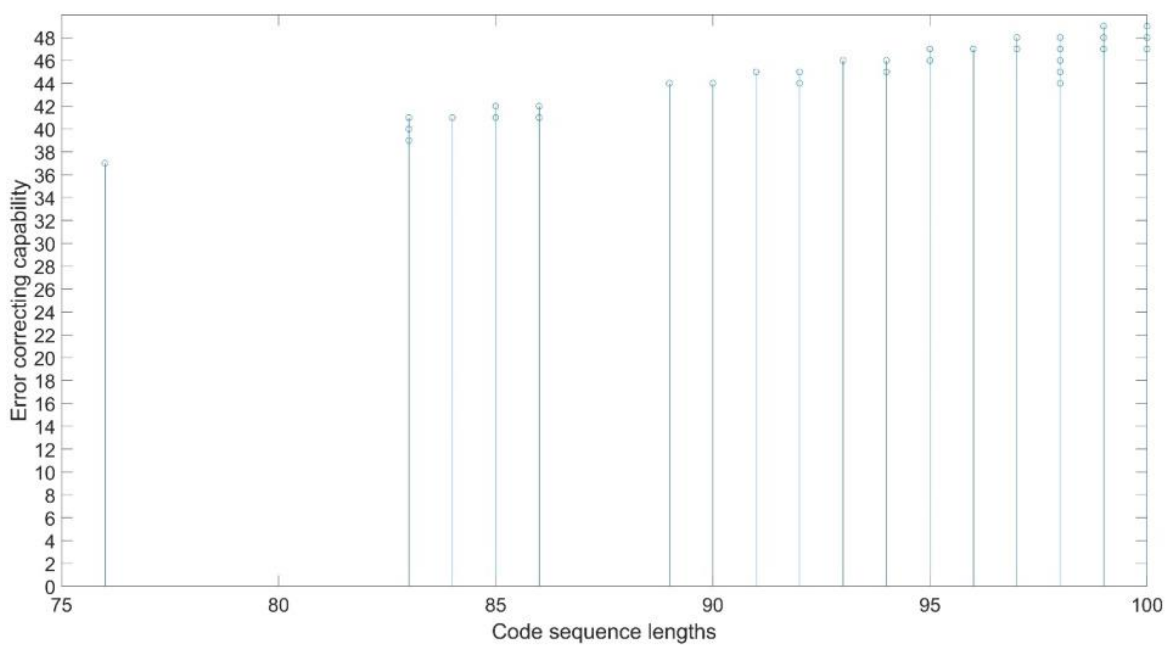

Figure 14
Set of optimal linear network code points for ρ = 1.1.

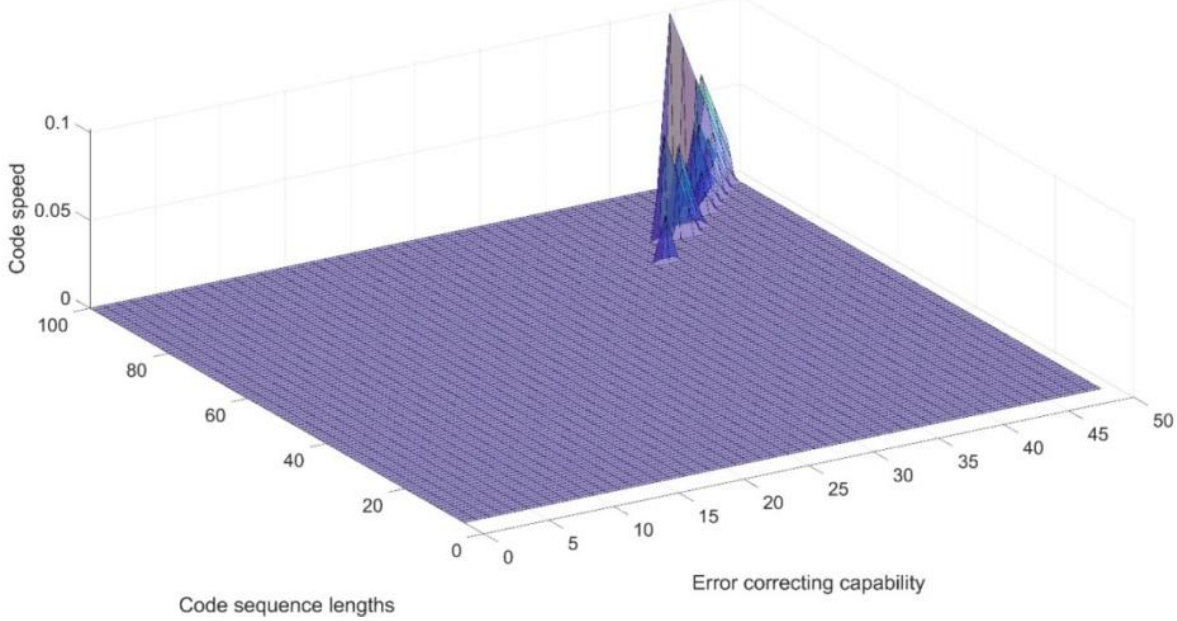

Figure 15
Graph of code speed versus code sequence length and code correcting ability, ρ = 1.1.

N	**T**	**R**
98	44	0.10
98	45	0,082
98	46	0.061
83	39	0.06
100	47	0.06

Table 5
Optimal points for ρ = 1.1.

Loss at ρ = 1.1 are so high that they are not acceptable for any class of data transfer, therefore, it is necessary to use coding with high corrective ability. Moreover, the choice of the parameters of the linear network code is very limited due to the presence of only 33 combinations, as we can see from the Figure 14 (all with high corrective power and a large code sequence length). The code speed is very small, which will lead to significant overhead both during transmission and during encoding / decoding operations. This fact requires improving the linear network coding method to minimize overhead, which will increase the code speed to an acceptable 0.5.

From the three considered cases, the workload of the queuing device shows that with an increase in the load on the “bottleneck”, a large correcting ability of the code is required. It allows smoothing out the correlated packet loss, but the increase in the length of the code sequence of packets is not noticeable. However, the number of variants of combinations (N,K) of the linear network code decreases with increasing length of the code sequence, which will require even longer combinations of the linear network code with a further increase in the load on the “bottleneck”. To transfer data of a small amount, amounting to several tens of packets, the use of coding will not have an effect, therefore systems with a re-request in this case have an advantage.

Conclusions

The study showed that the use of linear network code to combat packet loss due to network congestion eliminates the loss by introducing significant redundancy. This leads to an increase in overhead for encoding / decoding data and for the time it is assembled by the recipient. The solution to the problem may consist in the use of systems with retransmissions, as well as the optimization of linear network code to reduce the overhead of redundant data[11].

The sets of optimal values (N,K) of the linear network code obtained from the results of simulation modeling of congestion in networks with restrictions on throughput and packet processing time will find application in the message delivery protocol based on Automatic Repeat Request (ARQ) / Forward Error Correction (FEC) hybrid system[12]. The developed simulation model will clarify the use of ARQ and FEC modes.

Also, the developed simulation model will be useful for investigating denial of service attacks, as it allows evaluating the impact of an attacker on a network "bottleneck", changing the packet flow rate. For subsequent studies of the nature of losses and training of neural network models, the dumps obtained from the results of the work are very interesting.

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