ANALYSIS OF DEVELOPMENT TRENDS IN MODERN BUSINESS COMMUNICATIONS

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ABSTRACT

Advanced communication techniques are used to organize fixed-user access to corporate communication network (CCNs) resources, with the ability to quickly change the encoding speed during network overloads. Estimating the probability of a denial of service caused by a lack of free channel resources at Access Network was one of the research's aims. IP based network enable the delivery of different services. Primary determinants of users' subjective quality evaluation of communication: data transmission: fast, without errors or failures; video communication: high-quality images, no delays; audio communication: no voice distortion, no call failures. Important to network operators to make effective use of their own resources while maintaining transmission quality standards. Control over metrics such packet loss ratio, delay variations, and latency is crucial. For certain corporate networks, the implementation of 5G mobile communication equipment holds great potential, especially in terms of enabling remote access for staff members. Simultaneously, concerns over unauthorized access to information and the risk of packet loss during transmission have become increasingly pressing. This paper discusses the effectiveness of Low-Density Parity Check (LDPC) codes and polar codes in variable length packet transmission in a 5G system. The modeling results of 5G communication system operation at a fixed value of packet loss are discussed.

KEYWORDS: *corporate communications network, fixed access, denial of service, resource speed, mobile communication, packet loss probability.*

1 Introduction

IP based network enable the delivery of different services. Let us enumerate the primary determinants of users' subjective quality evaluation of communication: Data transmission: fast, without errors or failures; video communication: high-quality images, no delays; audio communication: no voice distortion, no call failures.

However, it's important to network operators to make effective use of their own resources while maintaining transmission quality standards. Control over metrics such packet loss ratio, delay variations, and latency is crucial.

The preferences of users and the capabilities of network operators frequently disagree. To some extent, packet transmission delays over a network are eliminated with the deployment of IP/MPLS switching technology combined with tunnel organization [1, 2]. However, errors in determining the correct amount may occur when utilizing a fixed tunnel speed resource, both greater and lower. A wide range of codecs is also available, and they vary from each other not just in terms of cost but also in terms of characteristics like connection speed and recovered signal quality.

Customer's selection of codecs can significantly affect such an indicator as the denial of service probability due to exhaustion of the allocated resource. It is of interest to evaluate how much this indicator depends on the codec speed and what the prospects are for using codecs with variable speeds [3].

The range of customer requirements is growing as a result of the potential for 5G mobile equipment: some applications demand peak data rates of several gigabits per second, while others need data packets to be delivered with a few milliseconds of latency, and still others need many years of battery life. The 5G technology network uses network function virtualization and network segmentation to meet these requirements. This creates a flexible underlying resource that can be easily modified to meet new requirements as they arise. The efficiency of promising codes can be compared by simulating the performance of a network access segment under fixed link quality metrics and, especially, under a given probability of packet loss [4, 5].

2 The organization of the variables affecting network capacity

The corporate communication network, or CCN, is a complex structure made up of many different components, including servers, computers, multipurpose devices often known as MFPs, digital PBXs, IP-PBXs, network adapters, switches, routers, system and application software, and cable systems (Fig. 1).

The purpose of creating a corporate network is to provide interconnection of system applications, which are located in different nodes, and to allow remote users to access them. Most often, a CCN is geographically distributed - it connects offices, branches, sites and other structures that are located at a considerable distance from each other.

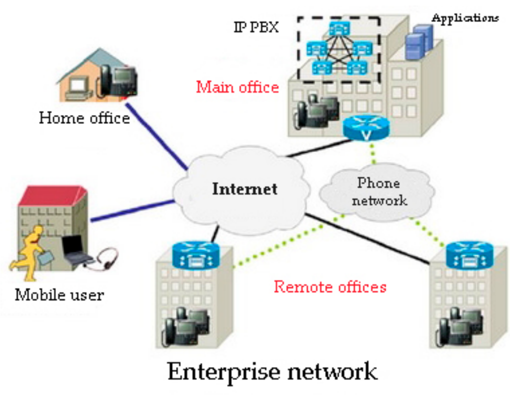

Fig. 1. Corporate communications network

Different network services react differently to different parameter changes. During periods of congestion (when the total number of transmitted packets approaches or exceeds the limit for a certain interface/channel), quality starts to degrade. Depending on the type of application, users of CCNs have different requirements. For example, a delay of up to 2 minutes in receiving a letter via corporate e-mail is acceptable from the user's point of view, but interruptions in communication, distortion, or loss of sound and image during videoconferencing sharply reduce the user's assessment of this service.

From the perspective of IP network operators, transport medium quality is determined by several parameters:

- connection speed to the backbone network;
- IP packet loss (loss percentage, Round-Trip Loss);
- Round-Trip Delay (Round-Trip Delay, RTD; Round-Trip Time RTT; Round-Trip Latency, RTL);
- delay variation (Jitter; IP Packet Delay Variation, IPDV; Packet Delay Variation, PDV). To evaluate service quality, the following parameters need to be monitored and controlled: delay, jitter, bandwidth, and packet loss parameters. Bandwidth requirements are set individually for each CCN project.

The large number of applications and the ever-increasing traffic lead to an increasing load on the corporate network. Increasing connection bandwidth and network device throughput (access switches, core switches, routers, multiplexers) have historically been used to manage network load.

However, these two types of traffic control do not take into consideration which type of traffic should be transmitted first and which next. The resource management system must take into account the application's QoS (quality of service) needs, the resources' availability, and the overall resource management strategy when distributing resources.

3 Evaluation of failure probability in provision of speech connection at the access network

The techniques used to design a CCN differ significantly from those used to design local area networks. The main difference is that geographically distributed networks connect objects that are remote from each other and often use relatively low-speed communication links. When designing an office local area network, the main costs are for purchasing equipment and cabling, but in geographically distributed networks, the significant costs are for the use of info-communication links. However, as business processes develop, the need to transmit more and more significant data traffic grows, and on the other hand, the price of renting resources from local operators is usually small but may increase. When designing the CCN, it is necessary to accurately evaluate the transmitted data size and to solve the issues of transmission rate rental at the access network (for example, Internet networks and specialized networks of MPLS technology).

The initial stage of the CCN design is especially important, since miscalculations and errors can later lead to the need to rent additional network resources and even to the creation of additional, parallel network structures.

Using one of the basic provisions of the mass service theory that the load intensity characterizes the average number of simultaneous connections on the network section, we define the average value of traffic intensity Qn (in this case, the necessary speed resource) as follows:

$$Qn = B1 * Y, \quad (1)$$

where value $B1$ characterizes codec speed per connection (Bit rate); Y – load intensity or average number of connections on the network section at the current moment of time.

Variations in traffic are not taken into consideration when estimating using the average value of Q, as demonstrated in formula (1).

Such properties as self-similarity of traffic flows and the characteristics of traffic flow intensity change over time [6-11] are not taken into account.

Its application may result in design errors related to the allocation of both too many and too few speed resources. In the first case, the probability of link failures increases, and in the second case, unnecessary costs are incurred in renting the resource. In addition, comparing protocols and technologies can lead to overly pessimistic or optimistic results.

It is suggested that estimations be made using the theory of mass service's mathematical tools. We may conclude, at least in part, from the statistics published in publications [3, 6-11], that the so-called "overhead" affect the throughput for a single connection, with the decrease in throughput being greatest when classic IP-codecs are used.

As a study topic, let's look at the access section that appears when an IP PBX is connected to a packet switching network in order to create a new voice connection. As an example, we may represent it as a mass service system of the format M/M/N/N, where the value N denotes the number of connections that can be realized on the speed resource Q allocated for these purposes.

This representation allows us to use the first Erlang formula to calculate the probability of call denial of service P, or to determine the acceptable load intensity Ya for a fixed value of Pa. Denial of service occurs when the entire speed resource available for a given type of traffic Q is fully utilized. The number of equivalent channels N will be determined by the formula

$$N = Q/q, \tag{2}$$

where the value q characterizes the average speed of one voice connection, taking into account the characteristics of codecs of different types and their percentage in the total overhead size of the transmitted packets

$$q = \sum_{j} B1j \times zj, \tag{3}$$

where j is an affect factor; zj is an estimate of the effect of j – factor on the value of q in relative units; B1j is the average speed of one voice connection taking into account the effect of j – factor (e.g., the use of a particular codec or a particular information security protocol).

Let us define the average number of simultaneous connections on the network section as Yai, defining it for a given value of call loss probability on the access section Pa due to network occupancy. We can obtain the allowed number of users in the i-th office as

$$Nai = Yai \,/\, (y1 * \mathrm{k}i), \tag{4}$$

where Yai is the allowed load intensity, determined by the first Erlang formula for a given value of the call loss probability on the access section Pa, and the number of equivalent channels N is determined by formula (2); $y1$ – specific subscriber load, E; Nai: allowed number of users in the i-th office; Ki is the percentage of calls routed to/from the packet network from other offices.

The value of N depends on the relationship between the devices with different types of codecs; besides, it is possible to change the principles of coding (transition in automatic mode to lower speeds) in case of overload. Table 3 shows the results of the calculations. The general algorithm for calculations is as follows:

1. Determine the set of affecting factors for a particular case and calculate the value of q according to formula (3).

2. Calculate the value of N by formula (2), based on the estimated values of the leased speed resource for voice traffic.

3. According to the 1st Erlang formula, using the Erlang calculator program, we obtain the function $Yai = F(N)$ at Pa = const. But first, we find the upper limit, $N\text{max} = Q/q$.

4. For the given values of *Yai*, we obtain the allowed number of users in the *i*-th office of *Nai*.

Depending on the goals and objectives, the order of calculations might be changed. For instance, a particular goal would be to determine the needed bandwidth for voice traffic at access section *Q* with a specific type of codec and a specified number of users.

Let's calculate in accordance with the proposed algorithm to determine the function *Nai* = *F*(*Q*). Let's assume that *Q* takes values *Q* = 2 Mbps – 5 Mbps. The G.711 codec is used in the office. The specific subscriber load *y*1 = 0.2 E. Percentage of calls routed to/from the packet network from other offices (*ki*) =0.3. There will be an error due to the fact that the values of N will be integers (rounding down). Table 1 shows the results of the calculations. Similar calculations performed for G.723.1 codec with an initial bit rate of 5.3 kbps are presented in Table 2. The possibilities of the G.722 codec to change the bit rate were taken into account, specifically Nai calculations for fixed *Q* value and Bit rate of 24, 32, 48, 56, and 64 kbps were performed (Table 3).

In view of this, comparing codecs based on access site bandwidth shows that, in the absence of any other factors, the IP codec offers a triple benefit in terms of user number.

The possibility of bandwidth adjustment by reducing the speed in the G.722 codec will have less effect than a complete switch to IP-telephony codecs, but on the other hand, it will allow to increase the number of users in a particular office more than twice with a fixed allowed call loss ratio.

Table 1

Calculation results of allowed Nai using G.711 codec with a bit rate of 64 kbps and MPLS protocol for initial data *y*1 = 0.2 E, *ki* = 0.3

Q Mbps	*Q* kbps	*N* MPLS	*Pa*, ppm	*Yai* MPLS	*Nai* MPLS
2	88.8	22	1	10.8	180
3	88,8	33	1	18,88	314
4	88,8	45	1	28,45	474
5	88,8	56	1	37,46	624
2	88.8	22	5	12.6	209
3	88,8	33	5	21,51	358
4	88,8	45	5	31,66	527
5	88,8	56	5	41,23	687

Table 2

Calculation results of the allowed number of users in the i-th office of the corporate network when using G.723. 1 codec with a bit rate of 5.3 kbps and using the MPLS protocol for input data (*y*1 = 0.2 E, *ki* = 0.3)

Q, Mbps	*q*, kbps	*N,* MPLS	*Pa,* ppm	*Yai,* MPLS	*Nai*, MPLS
2	32.6	61	1	41.6	693
3	32,6	92	1	68,2	1136
4	32.6	122	1	94,7	1578
5	32,6	153	1	122,7	2043
2	32.6	61	5	45.6	760
3	32.6	92	5	73,6	1225
4	32,6	122	5	101,2	1686
5	32,6	153	5	130,2	2169

Table 3

Calculation results of the allowed number of users in the i-th office of the corporate network when using G.722 codec and MPLS protocol for input data $y1$ = 0.2 E, ki = 0.3, Pa = 1 ppm

$B1$, kbps	q, kbps	Q, Mbps	N MPLS	Yai MPLS	Nai MPLS
64	88.8	2	22	10.8	180
56	80.8	2	24	12.2	203
48	72.8	2	27	14.4	240
32	56.8	2	35	20.5	341
24	48.8	2	40	24.4	406

3. Modeling results f 5G mobile access equipment performance in corporate communication networks

It is observed that there is a rapid deployment of 5G equipment solutions in the access section of enterprise communication networks. An important feature is that two new error-correcting channel codes are applied in 5G NR. In particular, Low Density Parity Check (LDPC) codes have replaced turbo codes for data channels, and polar codes have replaced Tail-Biting Convolutional Codes (TBCC) of control channels [4]. A message is a signal in a Physical Downlink Shared Channel (PDSCH).

To determine the input parameters for modeling, it was decided to refer to the parameters recorded in the 5G networks launched as of July 2022 and presented in the report [5]. Based on the results of average download speed of ten leading operators, the basic bandwidth target of 380 Mbit/s was established.

Another parameter that can be determined from open source data is the available spectrum band to the operator and used for the 5G NR network. Based on the data on commercially deployed 5G networks [5], a frequently used frequency band, band n78 (3.5 GHz), in which time-division duplexing (TDD) is supported, was selected for consideration.

At the physical layer, the amount of transmitted data is related to the transport block size (TBS). Note that TBS calculations according to the methodology set forth in recommendation TR 38.214 assume the use of LDPC code in the PDSCH channel. Transport block sizes, code rates and modulation orders for the polar code case were assumed to be equal to those determined for LDPC. Table 4 shows some parameters and transport block sizes selected for modeling.

Table 4

Parameters of calculations and sizes of transport blocks

Configuration	High-speed	Medium-speed	Low-speed
Modulation	256-QAM	16-QAM	QPSK
Code rate	948/1024	340/1024	251/1024
Number of allocated RB resource blocks	22	22	22
Transport Block Size (TBS)	25608	4480	1736

One of the limitations of using polar codes in 5G NR is the small size limit of the encoded block at the output of the encoder. In case of Downlink line, it is 512 bits (according to TS 38.212 section 5.3.1 [4]), and the maximum number of information symbols is 140 bits. A block transfer error rate (BLER) threshold of 10% was set for all configurations.

Simulation modeling was carried out in MATLAB 2022b environment, which provides wide possibilities for tuning the components of 5G NR standard signals. Figure 2 shows the block diagram of the used model and the main operations performed in it. The maximum values of the signal-to-noise ratio were preliminarily determined, above which the BLER was significantly less than the target value of 10%. Some results of modeling are presented in Figure 3.

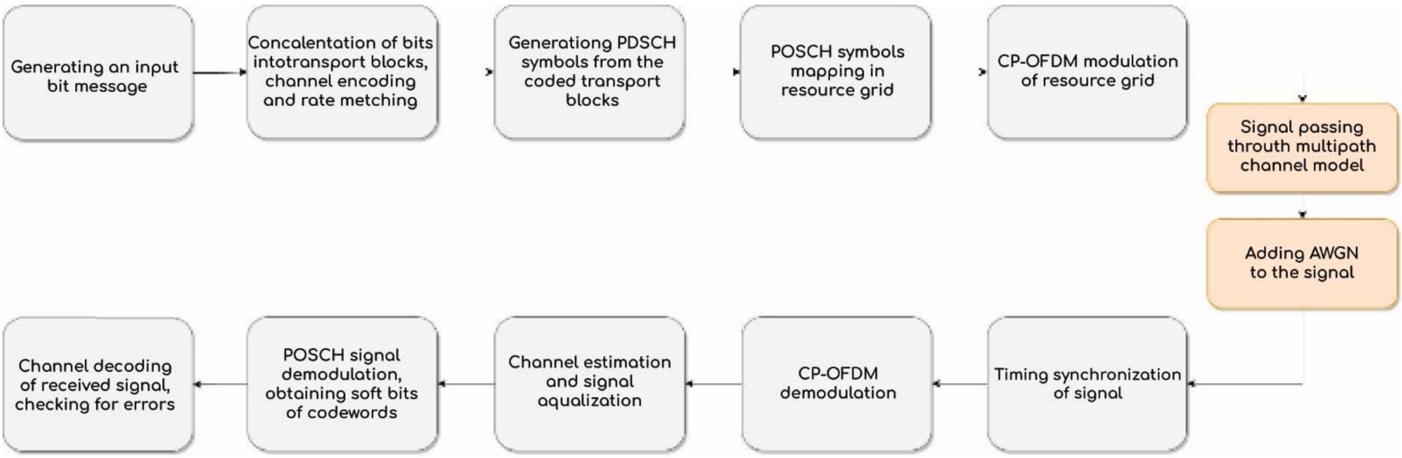

Figure 2. Block diagram of the simulation model

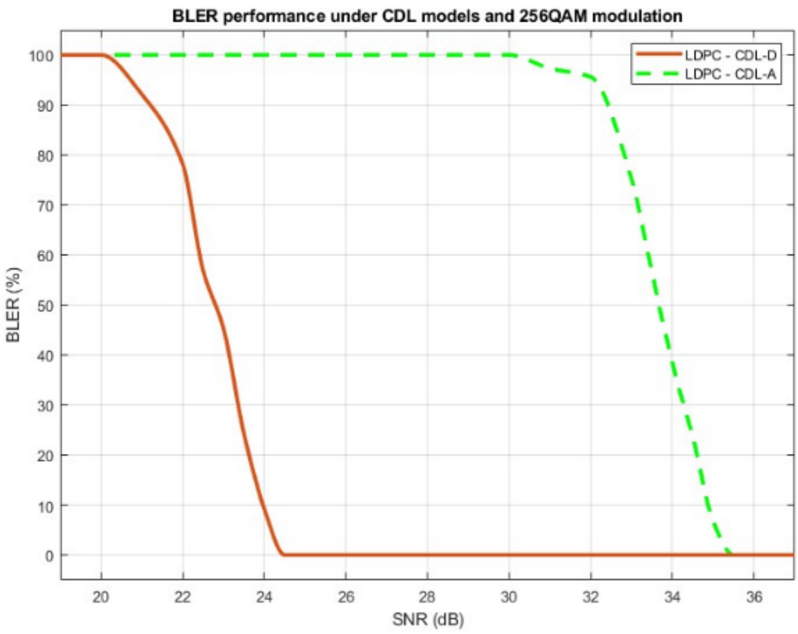

Figure 3. BLER dependencies on SNR signal-to-noise ratio for LDPC coding for high-speed configuration in CDL-D, CDL-A channels

Due to the large block size provided to a single device, for all channel models achieving the target BLER value of 10% required sufficiently high values of signal-to-noise ratio SNR. As a result of distortion, the resulting decoding error of a single bit compromises the integrity of the entire transport block, which is more critical for larger block sizes.

5 Conclusion

1. The apacity of the access sites of the CCN is affected by factors such as:

- the development of information encoding and decoding technologies (in particular, the transition to codecs of improved speech conversion quality, such as G.722);
- in the number of users and growth of traffic volumes in specific branches of the corporation;
- miscalculations in the allocation of the provided access speed both between offices and between applications and users of different classes of service (preferences in the use of terminal devices may change);
- redistribution of traffic flows between the company's branches when new offices appear or their location changes;
- growth of remote access traffic;
- change of generally accepted methods of traffic protection in packet networks (transition to the use of tunneling in combination with IPSec technology or MPLS technology).

2. The transition to tunneling principles in network structures such as IP-MPLS allows to increase security and reduce overall transmission delays, but increases the requirements for justification of the provided speed resource. The relevance of estimating the probability of failures due to unavailability of the bandwidth resource at the access site is related to the variety of codecs used and possible rapid growth of the number of employees in individual branches of companies. In fact, we are talking about a new factor of service failures, which should be taken into account. It is proposed to use Erlang models as a tool for calculating the allowed number of users in offices.

3. The simulation modeling of 5G generation mobile access system has shown that high modulation order (256-QAM) is one of the reasons leading to high SNR requirements for successful data transmission. Due to the small distance between modulation symbols in case of transmission interference, the probability of an incorrect reception and consequent incorrect decoding is high. It has been observed that polar codes standardized by 5G NR have a serious gain over LDPC 5G NR and, at the same time, a loss in decoding delay.

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