ASSESSING THE POSSIBILITIES OF IMPROVING COMMUNICATION QUALITY IN MODERN CORPORATE COMMUNICATION NETWORKS

Knaj Numa [1]

[1] Tartous University, Tartous, Syrian Arab Republic

ABSTRACT

The object of this study is an access node based on IP-PBX (Internet Protocol Private Branch Exchange) technology as part of a fixed network infrastructure. Unified communications is being implemented, facilitating comprehensive communication and collaboration. Specifically, it integrates voice, video, instant messaging, presence information, email, and document management onto a single platform. Companies across various sectors are investing in IP-PBXs to improve operational efficiency. IP-PBX telecommunications systems deliver voice and multimedia services over an Internet Protocol data network. They replace traditional circuit-switched and time-division multiplexing (TDM) PBX systems by using packet-switched networks and converging voice and data communications into a single infrastructure. This convergence offers significant advantages in terms of cost, scalability, management, and feature integration. Subject of the research: algorithms for distributing node bandwidth between serviced information flows. IP PBXs in corporate communications networks provide communications over existing data networks. Beyond simple cost savings, these systems lay the foundation for the development of unified communications (UC), which integrate disparate communication tools for services such as voice, video conferencing, instant messaging, and email into a single platform. This article presents an approach to studying the operation of a fixed-line node in the context of the transition to unified IP solutions. Various options for the combined transmission of traffic types such as voice and document flow files are discussed. A mechanism for priority resource allocation with a spillover effect is proposed. In this case, voice messages rejected in the main resource segment can be redirected (spilled) to an additional segment used together with file traffic. This approach reduces the overall probability of voice call loss. The negative impact of document flow will affect a small portion of voice traffic. The complex dynamics of flow interactions requires in-depth mathematical analysis.

KEYWORDS: *Internet Protocol; Private Branch Exchange; Complete Sharing; Overflow; Network Slicing.*

DOI: 10.36724/2664-066X-2025-11-6-23-33

Received: 15.09.2025

Accepted: 17.12.2025

Citation: Knaj Numa, "Assessing the Possibilities of Improving Communication Quality in Modern Corporate Communication Networks", *Synchroinfo Journal* **2025**, vol. 11, no. 6, pp. 23-33.

Licensee IRIS, Vienna, Austria.

This article is an open access article distributed under the terms and conditions of the Creative Commons Attribution (CC BY) license (https://creativecommons.org/licenses/by/4.0/).

Copyright: © 2025 by the authors.

1 Introduction

The object of this study is an access node based on IP-PBX (Internet Protocol Private Branch Exchange) technology as part of a fixed network infrastructure.

The global IP-PBX market is demonstrating robust growth. The broader "call management" market, which includes traditional and IP-PBX systems, was valued at $18.88 billion in 2023. It is projected to reach $48.11 billion by 2032, growing at a compound annual growth rate (CAGR) of 10.77%. Within this market, the IP extensions segment is considered the fastest-growing category. This growth is driven by increasing demand for cost-effective, flexible, and high-quality communications solutions, the widespread adoption of 5G technology, and the widespread bring-your-own-device (BYOD) trend.

Unified communications (UC) is being implemented, facilitating comprehensive communication and collaboration. Specifically, it integrates voice, video, instant messaging, presence information, email, and document management onto a single platform.

Companies across various sectors, particularly banking, financial services, retail, e-commerce, and hospitality, are investing in IP-PBXs to improve operational efficiency. IP-PBX telecommunications systems deliver voice and multimedia services over an Internet Protocol (IP) data network. They replace traditional circuit-switched and time-division multiplexing (TDM) PBX systems by using packet-switched networks and converging voice and data communications into a single infrastructure.

This convergence offers significant advantages in terms of cost, scalability, management, and feature integration.

The core network is a critical component of the architecture, not simply a data link. Quality of Service (QoS) is essential for prioritizing voice traffic over data packets to minimize latency, jitter, and packet loss, which directly impact call quality. This is implemented using mechanisms such as DiffServ (differentiated services) on network switches and routers

Traffic arriving at an IP PBX has a complex set of interrelated characteristics: heterogeneity – a combination of delay-sensitive voice and data traffic; randomness – flows are often modeled by Poisson processes; quality of service (QoS) requirements – voice communications require minimal delays and losses; limited access node bandwidth – efficient r source allocation between heterogeneous flows is essential [1-5].

Considering these characteristics is critical during design, for resource planning, and for ensuring guaranteed voice communication quality in the face of bandwidth competition. Otherwise, there is a risk of degraded voice connection quality, increased losses, and ineffective use of network resources, which directly impacts user satisfaction and the reliability of telecommunications services. Subject of the research: algorithms for distributing node bandwidth between serviced information flows.

2 Functional model of a fixed access node

A fixed access node based on IP-PBX is an integrated communication system, including a virtual PBX, a fixed access system, subscriber terminals and gateways for connecting to public networks (see Fig. 1). IP PBX performance is analyzed in terms of efficient processing and prioritization of mixed traffic (voice messages and files) under conditions of limited channel resources. IP PBX algorithms for resource allocation between priority voice messages and data files are studied.

IP-PBX (also known as IP-PBX) systems are based on unified software and standard multiservice platforms. IP-PBX systems transmit information over the public internet using IP-SEC (Internet Security) protocols or through specialized IP-MPLS (Multi Protocol Label Switching) networks.

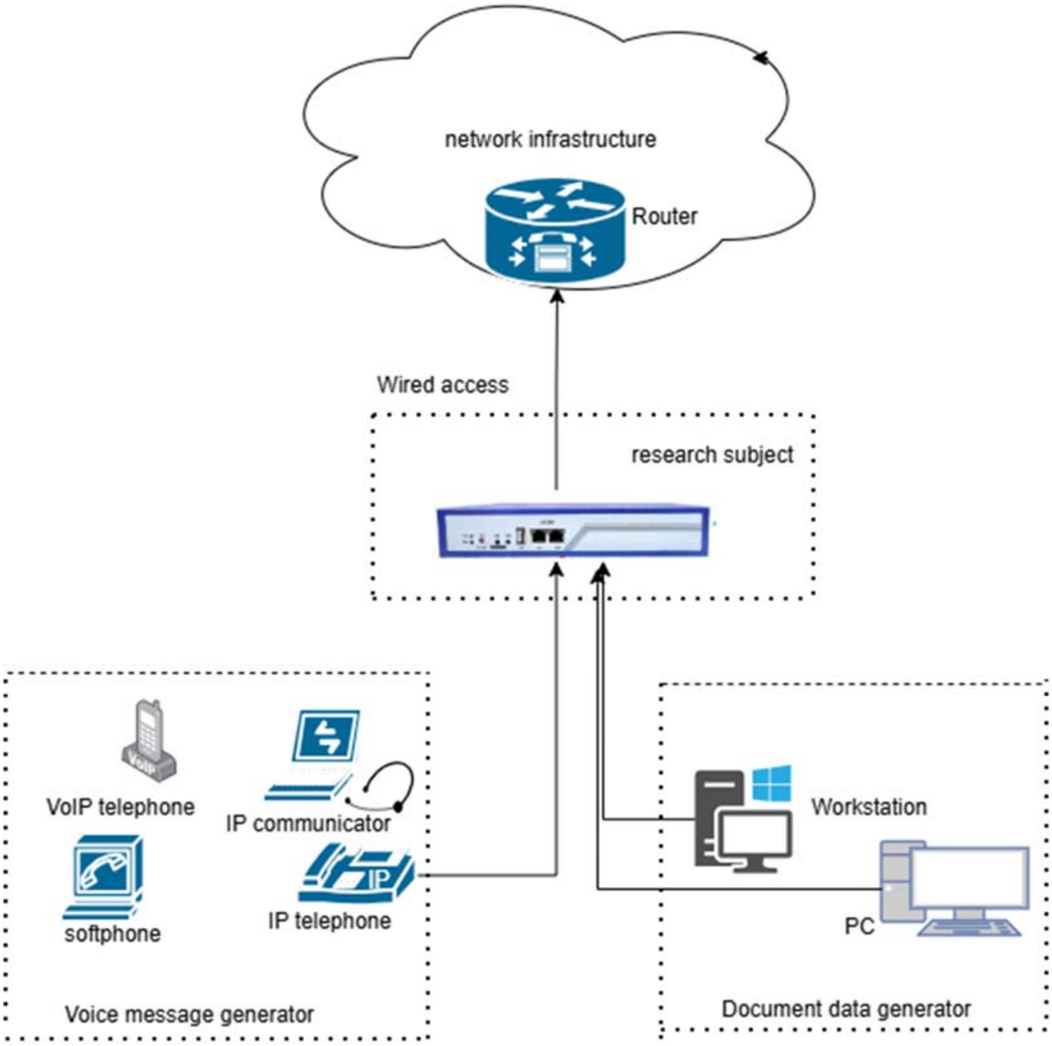

Figure 1. Including a fixed access node in a packet network

IP PBXs are widely used in corporate communications networks. Among the most in-demand features are the following [2, 3, 6-9]:

implementing a personalized approach to each client;

discovering effective service promotion methods.

A characteristic of the development of the computer industry, which competes with traditional communications systems, is the desire to use universal approaches. A server of any brand can be used with equipment from other brands, and any operating system can be used, be it Windows, Linux, or macOS. An example is the popular multiservice platform Asterisk, which is open source. Asterisk is flexible software that can be installed on any Linux-based platform. Besides Asterisk, there are competing VoIP PBXs, such as Elastix and Askozia PBX.

The most commonly used codecs in digital communication networks are G.711, which convert speech into a digital data stream with a primary digital channel speed of 64 kbps. Low-speed codecs of various types are used directly in IP telephony networks. Recently, support for wideband voice codecs such as G.722 has become popular.

They provide HD Voice (High-Definition Voice), i.e. high-definition sound, operating at a speed of 48, 56, or 64 kbps. Thus, the G.722.1 codec provides operation at speeds of 24 and 32 kbps with a bandwidth of 50 Hz – 7 kHz. A frequently used variant of the G.722.2 codec, also known as Adaptive Multi Rate – WideBand (AMR-WB), offers the ability to quickly change the compression rate as the bandwidth of the data transmission network changes. G.722.2 defines nine different transmission rate modes, ranging from 6.6 kbps to 23.85 kbps. In Russia, it is used in cellular operator networks under the name HD Voice technology.

A commonly used approach to assessing the throughput of packet networks and their components (including in the access subsystem) is to estimate the required speed per unidirectional connection. This is based on the size of the transmitted packets and the packetization period, which in turn depend significantly on the codecs and network protocols used, which generate and attach so-called "overhead" costs to each information packet. We will determine the network layer throughput for a single voice connection using the formula:

$$B = 2 \times (Pt \times 8 / 1000) \times N, \quad (1)$$

where Pt (Payload) is the final packet size, taking into account overhead, in bytes:

$$Pt = RP + O, \quad (2)$$

where RP is the packetization size in bytes per packet, defined as RP = (Tpacket / 1000) × (codec throughput × 1000 / 8); where Tpacket is the packetization period; O is the overhead, in bytes; N is the number of packets per second.

The coefficient 2 reflects the two-way nature of the connection [2, 3].

The packet rate per second (PPS) can be calculated by taking the reciprocal of the packetization period. The required speed for a single one-way internet connection varies significantly between codecs and depends on the network technologies used (whether using IP-SEC or MPLS).

For example, a delay of up to 2 minutes in receiving an internal email is perfectly acceptable from a user perspective, but connection interruptions, distortion, or loss of sound or video can cause a negative reaction.

3 Factors affecting voice traffic transmission quality in packet-switched networks

For telecom operators, it's important to maximize network resource utilization while maintaining the expected transmission quality. This involves monitoring: packet delays; delay variations; and packet loss.

The universal criterion for comparing voice transmission quality across different technologies is the subjective assessment method recommended by the International Telecommunication Union (ITU-T) [2, 7, 8, 10-12].

The assessment is performed using Quality Rating (R) on a 100-point scale or Mean Opinion Score (MOS) on a 5-point scale (see Table 1).

Table 1

Speech quality assessment according to ITU-T Recommendation G.109 (09/99)

Range R	Speech quality category MOS score	
90< R <100	best	4.34 – 4.5
80< R <90	high	4.03 – 4.34
70< R <80	medium	3.60 – 4.03
60< R <70	low	3.10 – 3.60
50< R <60	poor	2.58 – 3.10

MOS units are related to R by a complex nonlinear dependence (Rec. G.107). The highest quality R = 100 corresponds to MOS = 4.5. In practice, for a quick recalculation in the most important range 2.5 < MOS < 4.4, a linear approximation of the form MOS = R / 20 is convenient. Its error is less than 5%, which is quite acceptable, taking into account the spread in subjective assessment. Connections with MOS < 2.5 are not recommended. Moreover, for good quality connections, it is desirable to limit yourself to the first three speech quality categories from Table 1, that is, to ensure R> 70, or MOS> 3.5. ITU has developed a recommendation on the issue of standardizing quality of service in IP networks Y.1541 "Network Performance objectives for IP-based services". It defines QoS (quality of service) classes for each data flow between the user and the Internet access provider.

Quality of service standards are defined for the entire end-to-end section of the IP network architecture model (IP network Cloud). IP networks define six classes of service (from 0 to 5). Each class corresponds to specific IP network applications. Upper bounds for the following parameters are defined for each class of service: IPTD – IP packet transmission delay; IPDV – IP packet delay variation; IPLR – IP packet loss ratio; IPER – IP packet error rate.

IP-based networks not only deliver data but also serve as transport for a wide variety of services with varying network requirements. From the perspective of an IP network transporting network services, the quality of the transport environment is determined by several parameters:

- connection speed to the backbone network;
- IP packet loss (loss rate, or packet loss);
- round-trip delay (RTD; round-trip time RTT; round-trip latency, RTL);
- jitter (IP packet delay variation, or IPDV; packet delay variation, or PDV). As an example, consider jitter, which is defined in RFC 3393 as the difference in the end-to-end delays of two packets (Fig. 2).

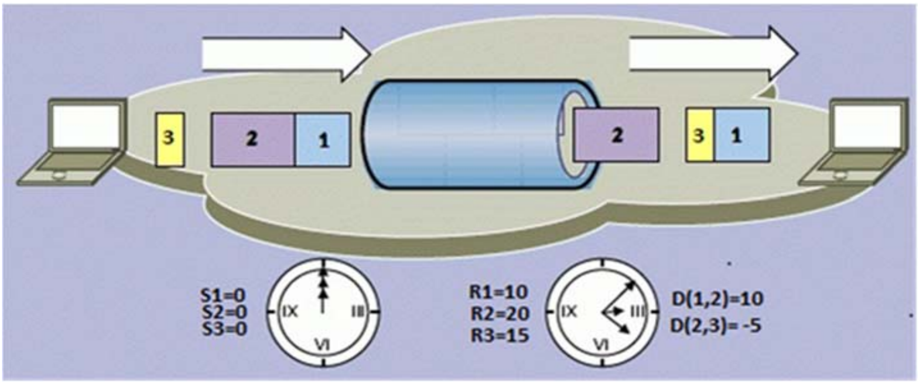

Figure 2. Delay Variation Due to Packet Delivery Variation

If R is the packet's sending time and S is its delivery time, then the PDV value for packets with numbers i and j is calculated as follows:

$$D_{i,j} = (R_j - R_i) - (S_j - S_i) = (R_j - S_j) - (R_i - S_i).$$

RFC 3550 defines a method for calculating the variance for a series of packets:

$$J_i = J_{i-1} + (|D_{i-1,i}| - J_{i-1})/16.$$

The measurement uses average PDV values over a given time period. Delay variations arise from the very nature of packet switching in an IP network. Ideally, the variation is zero, meaning packet delivery times are uniform.

However, due to the non-uniformity of the network flow passing through network nodes, as well as the operation of differentiated service mechanisms for network traffic, variations are not zero. It should be noted that the voice and data transmission quality indicators listed above characterize and normalize only a portion of the characteristics of packet traffic transmission.

4 Features of joint transmission of speech and document files

At the IP-PBX access point in a packet network, the connection speed is typically limited and is selected based on the cost of resource rental. When transmitting voice packets and document files simultaneously, an extremely unpleasant phenomenon occurs: voice packet transmission stops while the file is being downloaded. All active subscribers experience lengthy connection interruptions. This factor is not taken into account in ITU-T recommendations. But its negative impact is felt by users. There are several options for protecting against this phenomenon: transmitting data traffic in fragments, minimizing the mutual influence of different types of traffic using the Network Slicing approach.

Tables 2 and 3 present the calculated voice connection interruptions during document file transfers. Thus, the larger the file being transferred, the longer the connection interruption will last. Similarly, the greater the available bandwidth, the faster the file will be transferred. File transfer essentially consumes all available bandwidth for a certain period of time. Voice call packets can wait for transmission for a limited time. After this time, they are replaced by new ones, which may also be transmitted or lost.

Table 2

Calculated voice connection interruption duration in M/G/1 QSO for a 1 MB file size

Document file fragment size, MB			
1.0	0.5	0.2	0.1
Voice connection interruption duration per TF seconds during document file transfer in fragments			
2.67	1.34	0.534	0.267

Table 3

Calculated voice connection interruption probability in M/G/1 QSO per TF seconds for an initial file size of 1 MB for G.711 codecs at C = 6 Mbps

Number of users			
500	450	400	350
Probability of all voice connections being interrupted per TF seconds during document file transfers			
0.247	0.0200	0.0158	0.0121

The scientific objective of the study is to develop and analyze a mathematical model for flexible resource management in an access node based on an IP-PBX with heterogeneous traffic, which allows for increased efficiency in bandwidth utilization by routing excess priority traffic between logical segments (slices) of the network.

The wide variety of traffic management strategies can be summarized in three basic models, each with its own advantages and disadvantages [9, 12-15].

Complete Partitioning (CP) systems. In this model, a shared pool of N channels is rigidly divided into segments, each assigned to a specific type of traffic. In the context of our problem, this would mean allocating K channels exclusively to voice traffic and (N-K) channels to file traffic. As V.A. Naumov rightly notes, the main advantage of CP is the complete isolation of services, guaranteeing that "aggressive" file traffic cannot affect voice service.

However, this approach is extremely inefficient in terms of resource utilization. Under uneven load conditions, typical of real networks, a situation where all channels in one segment are occupied and failures occur, while in another there are available resources, is typical. This leads to excessive requirements for the overall capacity of the system to achieve the specified loss probabilities [4-6].

Complete Sharing (CS) systems. In this model, all N channels form a single pool, available for requests of any type. The CS strategy maximizes channel utilization, since any available resource can be immediately occupied by an incoming request. However, when servicing heterogeneous flows jointly, CS leads to "starvation" of low-intensity traffic with high QoS requirements. Consequently, with the CS strategy, file traffic can monopolize a large portion of the channels, leading to unacceptably high probabilities of voice message loss, which is unacceptable from a quality of service perspective.

To address the shortcomings of CS, priority-based service models were developed. The Russian scientific school has made significant contributions to this field. E.S. Kochkareva has thoroughly studied various priority-based service disciplines, including systems with absolute priority without interruption. In such systems, voice requests have unconditional priority when accessing free channels, and file requests are serviced only when there are no pending voice requests in the system. This effectively protects voice traffic, but leads to a sharp increase in file data transfer times during periods of high voice load. In the extreme case, with a constant voice traffic load close to unity, file traffic can be completely blocked [5, 6].

Recognition of the limitations of classical models has spurred the development of hybrid and adaptive approaches. The idea of dynamically redistributing resources between segments based on current load has been actively explored over the past decade.

In their paper, M.A. Snytkina and A.V. Gorelov propose an adaptive resource management algorithm for 5G networks that recalculates the boundaries between virtual segments in real time based on measured traffic intensity. The authors demonstrate a significant increase in utilization compared to CP. However, such systems are inherently slow and complex to implement, due to the need for continuous monitoring and decision-making [1, 2].

Closer to the topic of our research are models with so-called traffic "overflow." The concept of overflow was initially used in hierarchical communication networks, where calls that could not find a free channel in their zone were routed to neighboring ones [8-10]. In the context of multiservice networks, this idea has been transformed into a conditional access strategy, in which traffic from one class can access resources reserved for another class if certain conditions are met (for example, when the "native" segment is underutilized).

The priority-sharing model with spillover proposed in this study is an extension of these ideas. Its key difference is that two access levels are created for voice traffic: guaranteed (primary segment) and conditional (common segment). File traffic, meanwhile, constantly competes for resources in the common segment with spillover voice traffic. This structure allows for a targeted study of the tradeoff between voice service level and data service efficiency.

5 Network slicing

Modern communication networks are forced to handle a wide variety of data types simultaneously. Each type has its own stringent connection quality requirements. For example, voice messages dominate real-time traffic and are extremely sensitive to delays. At the same time, high throughput and guaranteed error-free delivery are more important for file transfer.

When all these heterogeneous data streams try to fit simultaneously into channels with limited bandwidth, they interfere with each other, and a fundamental and familiar problem arises: how to most efficiently distribute these resources among competing heterogeneous streams.

There is a traditional method for solving this problem called "Network Slicing."

This is an architectural approach in which a single physical network infrastructure is virtually divided into several isolated logical segments called "slices." Each slice is configured to efficiently handle a specific type of traffic with unique quality of service requirements. Real-time traffic (voice messages) is served in its own dedicated slice. The file request stream is served in a separate, independent slice (Fig. 3).

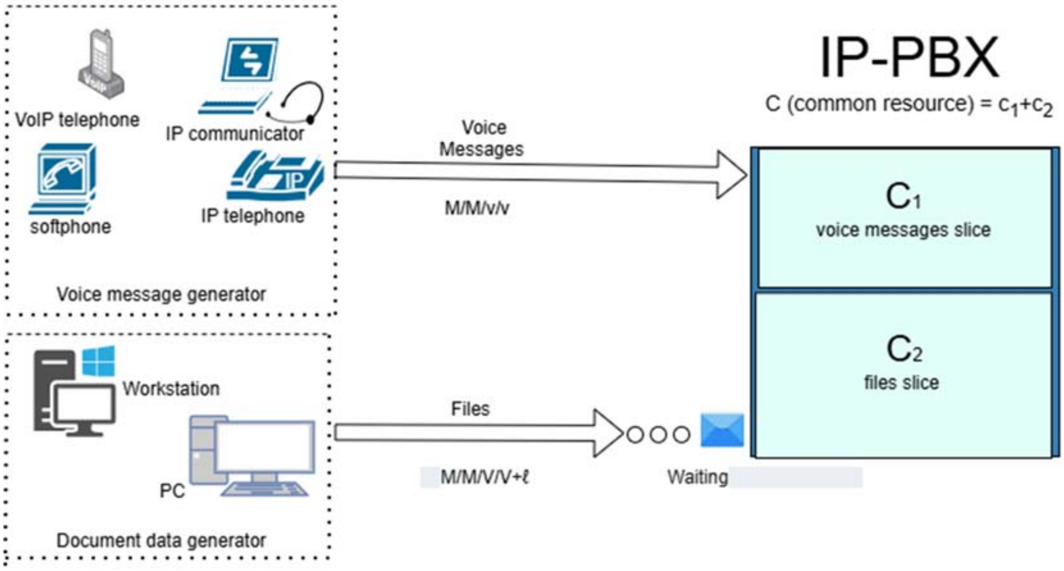

Figure 3. Illustration of the speed resource distribution using the "Network Slicing"

For static allocation, resources (virtual channels) are assigned to each slice on a permanent or long-term basis. Requests received in one slice cannot be redirected to another, even if there are available resources, which can lead to inefficient utilization if the load in one slice is low and in another high.

The model under study represents a more flexible resource sharing scheme. Server resources are explicitly divided into two parts, but there is interaction between them. The isolated slice is intended exclusively for real-time traffic, which is served in lossy mode. If all channels in this slice are busy when a voice request arrives, the request is not immediately rejected.

It should be noted that determining the number of virtual channels in the available speed resource involves using the characteristics of the specific codecs that are supposed to be used.

Redirection Mechanism: Instead of being rejected, voice requests are redirected as redundant requests to a shared resource slice designated for file traffic slice.

The shared slice is used for file traffic and redundant voice requests redirected from an isolated slice.

Files can wait in the queue, while voice messages have priority service.

The resources of the shared slice are shared between the two traffic types, improving overall utilization.

The system's structural diagram, shown in Figure 4, visualizes the problem under study. Streams from IP phones and softphones, representing voice traffic, are received for service. A key feature of the model is a priority resource sharing mechanism with spillover: voice messages that are rejected in the primary resource segment can be redirected (spilled) to an additional segment shared with file traffic. This approach reduces the overall probability of voice call loss, but creates complex flow interaction dynamics that require in-depth mathematical analysis.

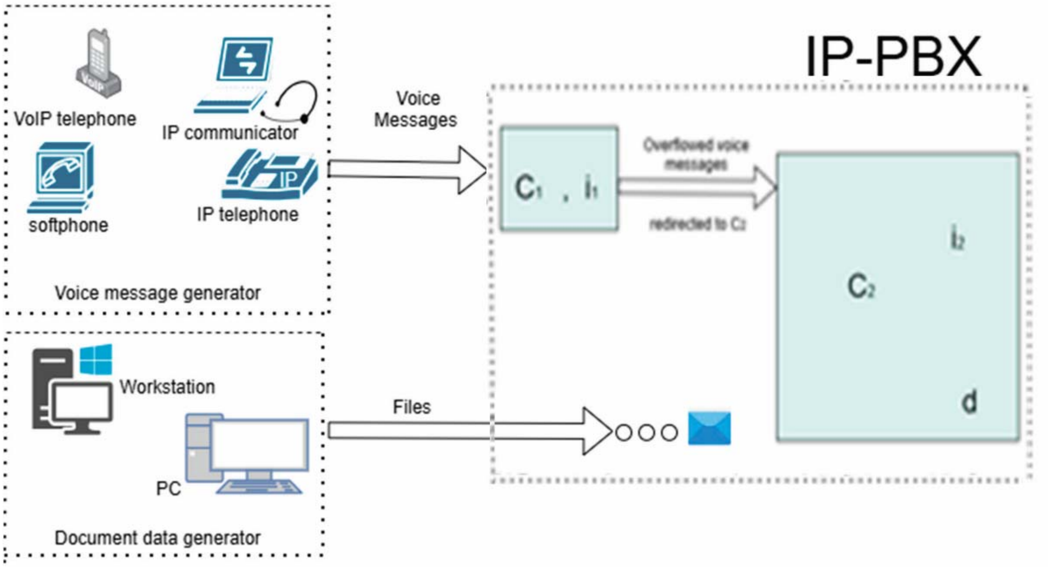

Figure 4. Flexible traffic distribution option between slices

Figure 5 shows the results of calculations of the probability of long interruptions in voice communication from the number of users for the slicing options in Figures 3 and 4.

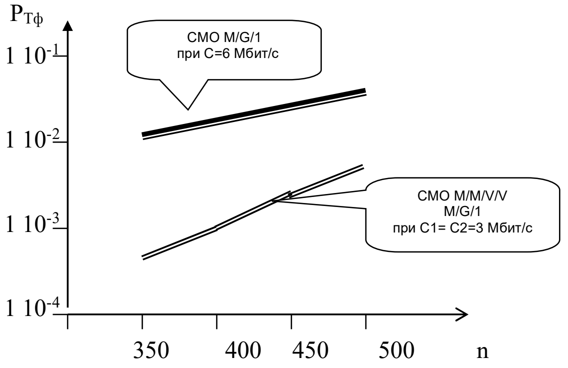

Figure 5. Dependence of the probability of voice communication interruptions per TF seconds on the number of users in an IP-PBX with G.711 codecs for document files of 1 MB, transmitted in whole or in fragments

The calculations confirm the potential of dividing the bandwidth resource on the PBX access section to Internet or IP-MPLS network resources into two streams: voice requests only and mixed traffic consisting of some voice requests (excessive for the first stream) and document files.

6 Round robin principle for document file fragmentation

The Round Robin (RR) principle is a fundamental approach to managing resource allocation in computing and network systems, particularly when it comes to multitasking and ensuring fair access to limited resources. The basic idea is to service all tasks or devices in a cyclical fashion, each allocated a time slice to perform the work or a virtual channel resource. When this principle is implemented in an IP-PBX, after a document file fragment is transferred, the access resource is passed to the next element in the queue.

In terms of its basic principles and mechanisms, Round Robin can be thought of as a circular queue in which each element is a process, a service request. In the classic understanding of RR, each task receives a fixed execution time, after which it is moved to the end of the queue. This approach ensures that no single device can dominate resource utilization, especially in network systems where devices may have different technical characteristics and operating speeds [2-4, 12-17].

The Round Robin principle has found application in the context of data transmission in the latest Wi-Fi standards and in LTE networks, where devices differ in standards and data rates. It allows for the equal distribution of available time and bandwidth among all participants. This is especially important when the network contains both high-performance devices and older devices operating at lower speeds. Without a fair resource distribution mechanism, such devices can slow down the entire system.

One of the key aspects of applying Round Robin in network systems is determining the optimal time slice (in our case, file fragment sizes). If the slice is too small, the system will frequently switch between devices (in our case, between files), which will lead to excessive overhead in managing these switches and, ultimately, can reduce overall performance. On the other hand, if the slice is too large, devices with lower speeds can occupy the channel for extended periods, degrading the quality of service for higher-performance devices.

Therefore, it is important to find a balance that ensures efficient resource allocation and minimal latency. An important aspect of the study is modeling the data transfer process between devices with different technical characteristics. For example, when transferring a 1 MB file between several devices operating under old and new standards, it is necessary to estimate the delays that may occur both with and without Round Robin. Modeling allows us to visualize and quantify how transfer time varies depending on file size and the number of active users.

Resource efficiency under various load levels is a key factor in selecting and configuring the Round Robin algorithm. Under low network load conditions, Round Robin can effectively distribute resources between devices, minimizing delays. However, under high load, especially with a large number of active users, difficulties may arise. Round Robin can be integrated with QoS to ensure fair and efficient distribution of network resources.

7 Conclusion

The scientific novelty of the research is as follows.

1. A new architecture for flexible network resource management at the access level is proposed, based on a modified Network Slicing concept with the ability to effectively redistribute traffic between logical segments.
2. An original mathematical model of the operation of an access node with heterogeneous traffic, based on the use of a multidimensional random process, is developed.
3. A comprehensive analytical apparatus is created for studying the steady-state operating modes of the system and evaluating the key performance indicators of servicing heterogeneous data flows.
4. An approach to determining the configuration parameters of individual network segments is substantiated, allowing for achieving a balance between quality of service requirements and bandwidth efficiency.

REFERENCES

[1] S.N. Stepanov, “Teletraffic Theory: Concepts, Models, and Applications,” Moscow: Goryachaya Liniya-Telecom, 2015. 868 p.

[2] I.V. Stepanova, “Management and Monitoring in Tele-communication Networks,” Moscow: EBS MTUCI. 2025. 112 p.

[3] I.V. Stepanova, N. Knaj, "Analysis of the capabilities of MPLS technology for managing traffic in communication networks," *T-Comm*, 2022. vol. 16, no. 5, pp. 63-68. DOI: 10.36724/2072-8735-2022-16-5-63-68.

[4] I.V. Stepanova, N.A. Tsygankov, A.P. Smirnov, "Trends in the development of mobile communications in the Russian federation to eliminate the digital divide," *I-methods*. 2025. Vol. 17. No. 2, p. 4.

[5] UC Today. The Unified Communications Market Boom: Stats, Trends, and Leading Vendors. 2025.
[6] Vonage. What Is IP PBX? Definition, Im-portance, Features and More). 2025. URL: https://www.vonage.com
[7] M.V. Zakrieva, Ya.R. Makarenko, G.V. Kolodeznaya, “Elimination of digital inequality. Problems and Perspectives.” *Innovative Infocommunications of the XXI Century.* Khabarovsk. 2024, pp. 124-128.
[8] N. Knaj, "Using MPLS Technology to solve BGP "blackhole" problem," *Synchroinfo Journal*. 2022. Vol. 8, No. 3, pp. 7-11. DOI: 10.36724/2664-066X-2022-8-3-7-11.
[9] K.W. Ross, “Multiservice Loss Models for Broadband Tele-communication Networks,” Springer, 1995. 343 p.
[10] Market Research Future. (2025). Call Control (PBX-IP PBX) Market Analysis, Size, Growth and Forecast – 2032. 2025. URL: https://www.marketresearchfuture.com
[11] MarkWide Research. IP PBX Market 2025-2034: Size, Share, Growth. 2025. URL: https://markwideresearch.com
[12] Research and Markets. IP PBX Market Size, Competitors, Trends & Forecast to 2030-2025. URL: https://www.researchandmarkets.com
[13] SNS Insider. Call Control (PBX-IP PBX) Market Size, Share Forecast 2032. 2024. URL: https://www.snsinsider.com
[14] R. Shoshan, Y. Dahan, D.B. Erez, and R. Confino-Cohen. From Conflict to Care – Telemedicine Utili-zation During Wartime: A Retrospective Cohort Study. (This indicates the use of digital communication, which often relies on IP-based systems). 2025.
[15] TAdviser. Unified Communications Global. (This provides data on the broader UC market which heavily influences IP-PBX adoption). 2025.
[16] I.V. Stepanova, N. Knaj, "Possibilities of the resource reservation protocol for increasing the capacity and reliability of traffic transmission between switching systems," *T-Comm*. 2023. vol. 17, no.2, pp. 49-55. DOI: 10.36724/2072-8735-2023-17-2-49-55.
[17] I.V. Stepanova, N. Knaj, "Building a combined code based on turbo code with additional bit encoding," *H&ES Research*. 2024. Vol. 16. No. 2, pp. 13-19. doi: 10.36724/2409-5419-2024-16-2-13-19.