FEATURES OF INTRODUCTION OF VoIP-TELEPHONY SERVICES

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The analysis of the peculiarities of the creation of voice transmission technology through packet networks is performed in the work, taking into account that the role of the PBX is performed by the virtualized cloud platform. The work is a review work and is devoted to the promotion of the use of modern cloud technologies within the concept of Industry 5.0.

The development of IP telephony has radically changed the speed of voice information. The transmission of voice information over an IP network imposes much stricter requirements on the network infrastructure than any other application. The degree of network availability, packet loss and time delays can have a significant impact on the quality of voice communication [1-2].

To eliminate the factors of negative impact on the network on the quality of communication, it is necessary not only to design the network itself, but also to regularly monitor its operation, traffic analysis and its modeling. Of course, the data network itself is not the only possible source of various communication defects, problems can also be caused by poor quality headsets, unprofessionally configured digital-analog gateways, and so on.

While traditional computer voice data networks were based on circuit-switched trunks for telephone traffic, voice packet-switched technology is now increasingly being used. This method of voice transmission is called IP-telephony or VoIP.

In recent years, the interest of various actors in the information services market (telecom operators, Internet providers, equipment manufacturers and users) has grown tremendously in this type of communication, due to the development of new standards and protocols, when IP-telephone conversation came close to quality to a telephone conversation on classic telephone networks. This is due to the fact that IP-telephony can significantly save the required bandwidth, which inevitably leads to lower tariffs, especially for long-distance and international telephone calls. Despite a number of problems with the introduction of new technology (the difficulty of ensuring high end-to-end telephone quality, incompatibility of equipment from different manufacturers, the high cost of hardware and software), IP-telephony (including web applications) allows users to offer completely new, impossible for traditional telephony services and applications [3].

The concept of IP telephony service provided by a virtual hosting provider implies not only the use of the Internet as a medium for voice transmission, but the use of the IP protocol and technologies that provide reliable and high quality voice information in packet switched networks. The lack of guaranteed quality of service in speech transmission is compensated by the emergence of technologies such as Multiprotocol Label Switching MPLS, Resource Reservation Protocol (RSVP), Differentiated Services (DiffServ) and others.

It should be noted that the standardization of voice technologies based on the TCP / IP stack and their support by the leaders of the packet telephony market ensure the compatibility of equipment from different manufacturers and allow creating systems in which calls from analog telephone connected to the router port to a personal computer. or from a personal computer to a PSTN number in three IP telephony scenarios.

So, the introduction of VoIP-telephony services allows to obtain both advantages in the quality of telephony services and to ensure the appropriate level of cybersecurity when transmitting voice data in public networks.

References:

1. Поповский В.В. Определение частот формант голосового сигнала на выходе гребенки полосовых фильтров / В.В. Поповский, С.М. Бобрицкий, В.Е. Саваневич, В.Н. Ткачёв // Системы обработки информации. – Харьков: ХУПС, 2010. – № 5(86). – 121-124 с.

2. Бобрицкий С.М. Обнаружение интервалов группировки потерь речевых пакетов в пакетных сетях с использованием кодеков с подавлением пауз речи и компенсацией потерь / С.М. Бобрицкий, А.В. Колтыков, В.Н. Ткачёв // Системи озброєння та військова техніка. – 2011. – № 2(26). – С. 109-113.

3. Tkachov V.M. Architecture of Overlay Network with Nested VPN Tunneling / V. Tkachov, M. Bondarenko, M. Hunko // Сучасні напрями розвитку інформаційно-комунікаційних технологій та засобів управління: матеріали десятої міжнародної науково-технічної конференції. – Баку: ВА ЗС АР; Харків: ДП «ХНДІ ТМ»; Жиліна: УмЖ, 2020. – С. 36.