Performance of VoIP Services with Integrated Analog Peripherals

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Abstract – **In this paper, the performance of VoIP services with integrated analog telephones is determined. The integration of analog phones was carried out with the analog telephone adapter Grandstream HT702. By measuring the value of key QoS parameters of conversational speech realized through the IP PBX AXON switch, the performance of VoIP services is determined. Packet delay, as the most important QoS parameter of conversational speech, was within the given limits. The package loss was also within the given limits, while the mean value of the jitter oscillated around the allowed value. Jitter was successfully compensated by the receiving buffers and had no influence on the conversation speech. Based on the experimental results, it was found that the key QoS parameters defined for the conversation speech were satisfied, which confirms the successful integration of analog telephones into the VoIP service.**

Keywords – ***VoIP, IP PBX, AXON, ATA, QoS*.**

I. Introduction

The great expansion and distribution of computer networks made it a very suitable medium for the transmission of various content. In addition to the exchange of textual documents in electronic form, computer networks have become the basic medium for the distribution of multimedia content [1]. Regardless of the fact that they are not designed for this purpose, with increasing network flow and the emergence of modern network communication protocols this service has become possible [2]. In order to offer satisfactory resources to different network applications, it is necessary to classify them according to their needs. Standardization of network applications has been carried out by international standardization bodies such as IETF (*Internet Engineering Task Force*) and ITU (*International Telecommunication Union*). From the point of view of users, network applications can be divided into those tolerant to packet errors (multimedia applications), and those tolerant to packet delay (Web search or e-commerce applications) [3]. Multimedia network applications are further classified into three basic classes. Network applications that share stored audio or video content in real time are classified in the first class. The second class includes applications that support conversation and video telephony. Network applications that allow streaming of audio and video content streaming are classified in the third class. This application class is similar to classic radio and television diffusion. The basic problem with all applications for distributing multimedia content through computer networks is the realization of the appropriate QoS (*Quality of Service*) [4]. The origin of this problem is in the concept of computer networks based on TCP / IP protocols that do not guarantee QoS, but only "best effort" QoS services are provided. This means that there is no guarantee that the package will be delivered at all, and if it is delivered, its timeliness and order are not guaranteed. For each of the classes of multimedia network applications, specific QoS parameters are defined. The key standardized QoS parameters of multimedia network applications are packet delay, packet delay variation, and packet loss. The realization of conversational speech through computer networks, or the Internet, is called Internet telephony or VoIP (*Voice over Internet Protocol*) [5]. From a user perspective, VoIP is similar to a traditional analog telephone service that is realized by circuit switching. Analog peripheral devices characteristic of a traditional phone service are still present in homes and small business environments. In order to take advantage of the existing analog telephone service infrastructure, technology for the integration of analog telephone devices in VoIP has been developed. Multimedia applications belong to the class of network applications that are tolerant of errors, but they are extremely intolerant to the package delay. Thus, for a conversation speech, the delay of 150 ms is allowed, but the delay of 150 ms to 400 ms can be tolerated. The acceptable variation of the delay is 1 ms, and the allowed PLR (*Packet Lost Rate*) is 3% [4].

A. State of Art

Getting the certain QoS in IP network is crucial and is usually achieved by setting up access and service policies [6]. Realization of QoS in IP VPN networks must provide the recommended lower limits for satisfying QoS requests at the network node level [7]. Enabling multilayer QoS in VoIP calls requires speed adjustment and resource management to integrate SIP and QoS mechanisms to wireless networks [8]. Different queueing policies have been considered to achieve the end-to-end QoS requirements for different types of network trafic [9]. In this paper, an analog phone is integrated into the digital IP PBX (*Internet Protocol Private Branch Exchange*) [10]. The integration of the analog phone into the IP PBX provides, among other things, additional features to an analog peripheral device such as voicemail, call waiting, call hold, personalization of ring tones, and so on. Integration of analog phones into IP PBX requires a set of time-consuming signal and protocol conversions so that QoS satisfaction can be critical.

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This paper discusses the performance of VoIP services in which analog peripherals ATA (*Analog Telephone Adapter*) are integrated. In the experimental part of the work, the Grandstream HT702 ATA device was used [11], which allows the analog telephone to be connected to the digital IP network. The Grandstream HT702 conducts the conversion of one communication protocol (analog) to another (digital) and vice versa. In the terminology of the ITU-T ATA, the Grandstream HT702 is a gateway that provides two-way communication in real time. The measurement of the parameters of the VoIP application was carried out in the AXON IP PBX realized in the local computer network. The AXON IP PBX central software phones are directly integrated through the network interface, while the integration of analog phones uses the Grandstream HT702 ATA device. The key QoS parameters for the conversational speech connections made between all terminal devices are measured. For this work, it is important to analyze the QoS parameters lured by communication between software IP phones and analog telephone devices. Using the open-source *Wireshark* software package, measurements were made and the results obtained were presented.

The second section describes the need to integrate analog telephone devices into the IP PBX central. A basic overview of the basic characteristics of the AXON central is given, its web interface is described and the basic characteristics of the used ATA Grandstream HT702 are shown. The third section describes the test environment and shows the obtained QoS results. The experimental results are compared with the target values of QoS and certain performance of VoIP services within the AXON IP PBX. In the fourth section, appropriate conclusions were drawn based on the experiments carried out and certain recommendations were given.

II. INTEGRATION OF ANALOG TERMINAL DEVICES IN IP PBX

The integration of analog terminal devices - analog phones or FAX devices - provides several business benefits to the IP PBX network platform. The basic benefit of integrating multiple communication networks into one is to reduce costs. Combining multiple business communications networks into one convergent network infrastructure greatly reduces capital costs, cabling costs, and operational costs associated with managing separate network infrastructures. New business opportunities by combining network data with voice calls are obtained. In this way, an integrated information exchange system is formed which saves time and enables more efficient decision making. The telecommunications industry has many years of experience in IP voice integration in traditional analog networks. It is estimated that organizations save 30% to 40% on costs associated with long distance communications [12]. In addition, significant savings are realized on administrative costs and costs related to maintenance of equipment. Integrated network services use a single network architecture to simplify system maintenance and upgrading. From the standpoint of modern users, the characteristics of analog terminal devices are very limited. In order to improve the functionality of analog terminal devices - standard analog telephone devices, and at the same time utilize the existing network infrastructure, they can be integrated into digital PBX telephone exchanges. In the telecommunications industry, the term IP PBX is a private telephone exchange using a TCP/IP protocol for managing telephone calls through a computer network.

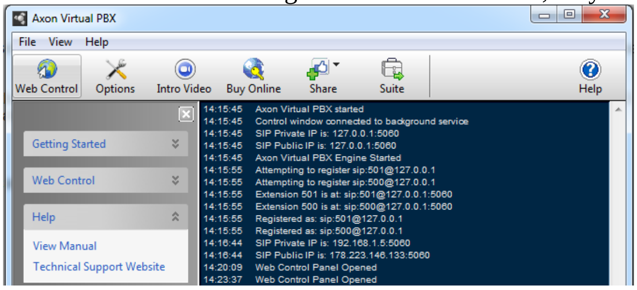

Fig. 1. Web interface of Axon IP PBX

The integration of the analog phone into the IP PBX provides additional features to an analog terminal device that is implicit in digital exchanges. Some standard IP phone functions cannot be easily implemented, while others can not be implemented at all because of the hardware limitations of the analog phones. IP PBX enables multimedia communication as well as the communication of sending simultaneous messages between two or more participants over the Internet or other compatible networks. They are mostly used by small and medium-sized enterprises, but they can also be found in large multinational corporations. IP PBX can exist as a physical hardware or just as a virtual software realization. In addition, it is possible to connect to the public telephone network PSTN.

In this paper the IP PBX of the company "*NCH Software*" was used [13]. Axon is a virtual IP PBX software designed to manage phone calls in a business environment or at a call center. AXON IP PBX is installed on a PC and supports up to 64 phone lines, as well as an unlimited number of extensions. AXON IP PBX works as a fully equipped telephone exchange that connects telephone lines and extensions using state-of-the-art VoIP technology. AXON offers all the usual features of a traditional PBX as well as routing all calls within the company. For the setting and controlling of the AXON switch, the Web interface shown in Fig. 1 is realized.

B. Analog Telephone Adapter Grandstream HT702

In order to integrate analog terminal devices into the digital IP network, the Grandstream HT702 analog telephone adapter (*Analog Telephone Adapter*) (Fig. 2) is used in this work. Analog telephone adapters are devices for connecting traditional analog phones, fax machines and similar user devices to a digital telephone system, or to a VoIP telephone network. An analog telephone adapter's digital interface usually

Fig. 2. Interfaces of ATA HT702 for connection from one side to the LAN, and on the other side, to analog terminal devices.

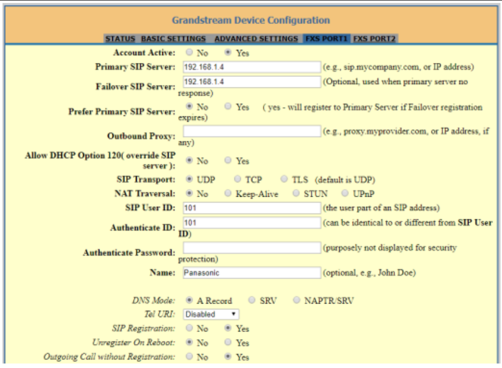

Fig. 3. Example of configuring the ATA HT702 FXS port 1

contains an Ethernet port for connecting to an IP LAN (Fig. 2). ATA communicates with the server (in our case with the AXON IP PBX) using one of the protocols H.323, SIP, MGCP (*Media Gateway Control Protocol*), SCCP (*Signaling Connection Control Part*) or IAX (*Inter Asterisk eXchange*). When an ATA is used to integrate an analog telephone, it encodes / decodes audio signals using one of the standard voice codecs such as G.711, G.729, GSM (*Global System for Mobile Communications*) and iLBC (*Internet Low Bitrate Codec*). The HT702 supports SIP signaling protocols for the VoIP network used in the experimental work. The ATA Grandstream HT702 is fully compatible with the industry-standard SIP standard and works with all compatible devices. Some of the features of the ATA HT702 are comprehensive support in the form of a callout for outgoing calls, G.168 echo cancellation, VAD (*Voice Activation Detection*), CNG (*Comfort Noise Generation*), hiding packet loss PLC (*Packet Loss Concealment*), etc. The HT702 has two RJ-11 FXS (*External eXchange Station*) ports for connecting to analog terminal devices such as analog phones or FAX machines (Fig. 2). ATA communicates directly with AXON IP PBX, and does not require additional hardware or any additional software. The Web interface for setting and managing ATA and a SIP environment is shown in Fig. 3.

III. Experimental environments

The experimental part consists in establishing telephone connections between software and integrated analog phones in the LAN through the AXON IP PBX [14]. A LAN is formed around a router that connects the wired and wireless part of the network as shown in Fig. 4. Measured QoS parameters for all performed experiments are shown in Tab I. QoS parameters are named according to [4]. The open source software "Wireshark" was used to measure the QoS parameters of realised VoIP traffic on the network. The DHCP (Dynamic Host Configuration Protocol) server is activated on the router so that all network components are assigned the appropriate IP addresses. Most available ATAs on the market have similar characteristics. The HT 702 model of Grandstream was available to us. Grandstream HT702 is assigned an IP address through which it can communicate with other network components. An analog telephone device is connected to one FXS port of the ATA HT702 device while the analog cordless telephone is connected to another. On FXS ports can be connected both analog dial phones and analog phones with

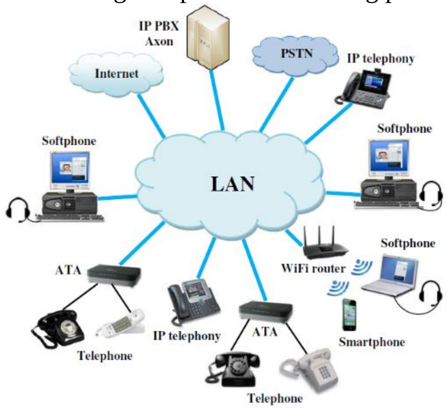

Fig. 4. Axon IP PBX with integrated analog telephones by ATA Grandstream HT702

DTMF dialing. An analog FAX machine can be connected to one of the FXS ports. All computers in the network are running Windows, and an AXON IP PBX telephone exchange is installed on one PC. All analog and software telephones from a local computer network are registered at the AXON telephone exchange before making telephone connections as required by the SIP protocol. NCH Software's Express Talk software phones have been installed on all network scanners so each of them can be used as a terminal device. In fact, each one of them can start and accept the call. Also, from each analog phone, it can be established, that is, accept a telephone connection with software phones. Fig. 4 shows a local computer network where the experimental part of the work is performed. Fig. 5 shows the interaction between the analog and the software telephone during the establishment, duration, and termination of the connection to the already known - received IP address (the phase of receiving the caller's IP address is not shown in Fig. 5). The provision of caller IP addresses is the task of the SIP protocol, which is used in the experimental part of this paper. In this case, the AXON IP PBX has the role of a SIP proxy that provides the current IP addresses of the users. Before any connection is established, it is necessary to perform SIP registration on AXON IP PBX of all software or integrated analog phones. Before any connection is established, it is necessary to perform SIP registration on AXON IP PBX of all software or integrated analog phones.

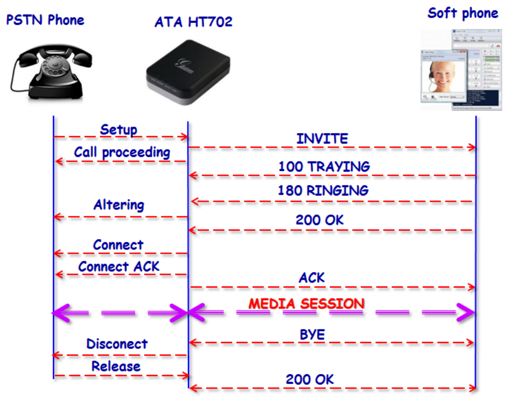

Fig. 5. Establishing a call to the known IP address between the analog phones integrated with the ATA H702 in the VoIP system.

TABLE I
EXPERIMENTAL RESULTS OF KEY QOS PARAMETERS

No	MEDIUM	APPLICATION	DEGREE OF SYMMETRY	KEY PERFORMANCE PARAMETERS AND TARGET VALUES						
				ONE-WAY DELAY (ms)		DELAY VARIATION (ms)			INFORMATION LOSS (%)	
				MEASURED	TARGET	MAX	MEAN	TARGET	MEASURED	TARGET
1	Audio	Convers. voice	Two-way	30.649	< 150	1.481	0.232	< 1	0	< 3
				129.319		7.135	1.852		0	
2	Audio	Convers. voice	Two-way	129.440		8.291	1.521		0	
				40.057		6.019	0.802		0	
3	Audio	Convers. voice	Two-way	55.775		5.537	1.057		0.2	
				40.796		5.012	1.158		0	
4	Audio	Convers. voice	Two-way	47.453		6.947	2.470		0	
				64.366		5.284	0.977		0.6	
5	Audio	Convers. voice	Two-way	26.501		1.286	0.176		0	
				139.625		11.742	1.357		0	
6	Audio	Convers. voice	Two-way	131.087		7.254	1.232		0	
				26.462		0.997	0.126		0	

After a successful SIP registration, connections can be established. Tab. I shows the results of key QoS parameters for six realized telephone connections. All connections have been made on one side of analog phones, and on the other side of software phones from LAN. In all experiments, it was a two-way conversation that formed two audio streams. Connections 1, 3, and 5 were initiated by analog phones integrated into VoIP with ATA Grandstream HT702, and accepted by software phones installed on computers from LAN. Software phones initiated the other three connections, and the integrated analog phones were accepted. Measured key QoS parameters that relate to one-way delay and loss of packets in all cases meet the prescribed ranges in [3] and [4]. This standard prescribed that the delay in one direction must not exceed 150 ms, which is also satisfied in all connections. The allowed loss of the package is 3%, so by insight into Tab. 1 can determine that this QoS parameter is satisfied. The variation of the delay (maximum and average) is allowed to be less than 1 ms. In some connections; considerably larger peaks are measured, while the mean value of this QoS parameter oscillates around the allowed. It should be noted that the variation of the delay did not significantly affect the quality of the speech, so it did not influence the conversation. The variation of the delay is simply compensated by the use of the receiving buffer.

IV. CONCLUSION

In this work, the integration of analog telephone devices into the VoIP system was made using the ATA Grandstream HT702. Checking of the realized QoS parameters of VoIP services of integrated analog phones was performed in the AXON IP PBX. Several bi-directional communication links between software phones and integrated analog phones were realized and key QoS parameters were measured. Based on experimental results, it was found that the key QoS parameters defined for the conversational speech were met. Packet delay, the most important QoS parameter for the conversion speech application, was within the given limits. Also, the loss of the package was within the given limits, while the variation of the case-by-case variation oscillated around the allowed value. The receiving buffers compensated the value of the jitter so that it did not have an effect on the conversation speech. Experimental results show that successful integration of analog terminal devices into a VoIP system based on the AXON IP PBX has been performed.

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