The Enterprise Telecommunication Network Design and its Implementation Using Technology PLC

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Abstract— This paper is focused on an enterprise telecommunication network design and its implementation using technology PLC (Power Line Communication) with ability of VoIP deployment. The design consists of two primary segments. Firstly, enterprise telecommunication network with software PBX based on Asterisk platform and secondly, telecommunication network with hardware IP PBX Panasonic KX-NS500, both networks based on backbone PLC technology. Important part of the solution is also implementation of connection of software Asterisk PBX and IP hardware Panasonic PBX KX/NS500.

Keywords-PBX, PLC technology, VoIP, Asterisk, network design;

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I. INTRODUCTION

Each company needs its own telecommunication system for providing internal and external communication. When the first public switched telephone networks were forming in 19th century, requirement for private telephone network arised. The telephone networks had been connection-oriented for many years. VoIP technology was developed in the 90s of the last century. It was based on an idea of voice transmission through the existing system of the Internet. VoIP technology brought many benefits mostly in economic point of view. A significant progress of transport technology PLC occurred when the VoIP was developed. It uses electrical wiring for information transmission. The technology is described in detail in papers [1, 2, 3, 4]. An example of comprehensive PLC system utilization is given in Fig. 1.
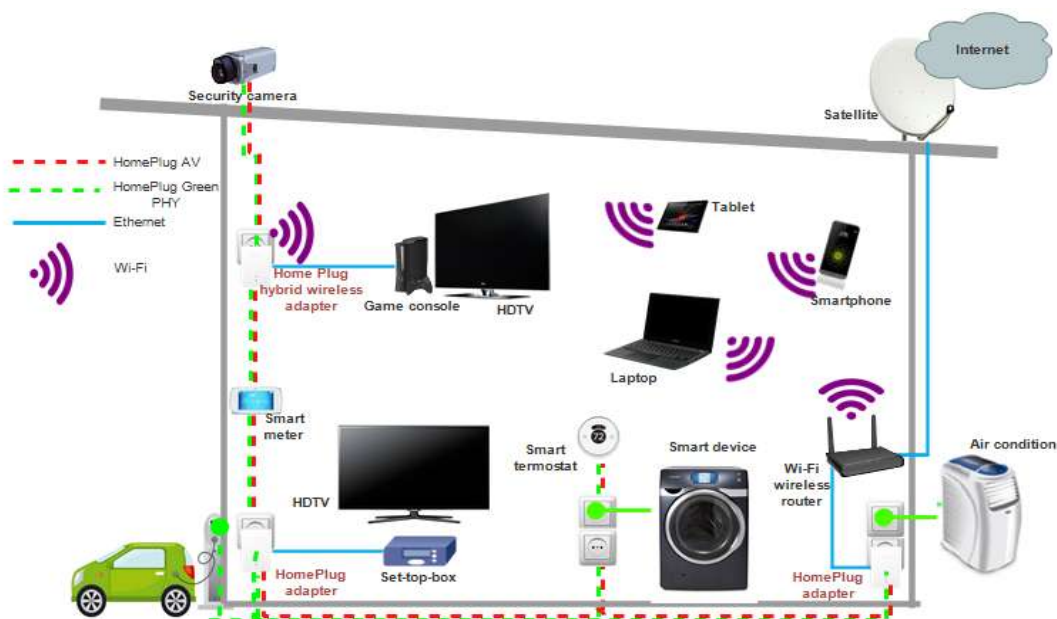

Figure 1. Example of comprehensive PLC system utilization.

II.   TELECOMMUNICATION PLC NETWORK DESIGN WITH SOFTWARE ASTRISK PBX

The enterprise telecommunication network segment based on PLC platform, using software Asterisk PBX is described in the first part of the paper. Considering a character of physical media which is electrical wiring, it was necessary to use PLC modems.

Modems Corinex HD-200-W2HF (Fig. 2) are older types of PLC modems and their transmission speed is up to 200 Mbps. Modems have network interface *10/100BaseT Fast Ethernet*, work in frequency range from 2 to 30 MHz and support standards IEEE 802.3 a 802.1 so they support Ethernet frames transmission.

Figure 2. Modem Corinex HD-200-W2HF.

Software Asterisk PBX design came from an open source solution. Asterisk works on Linux. It dispose of all the functionalities that are provided by hardware PBXs. Asterisk platform is a very flexible solution of PBX which is result of extension modules connection ability. Primary ways of its usage are for example VoIP gateway for various protocols (SIP, IAX, H.323), Voicemail service, Voice Guidance Server, IVR and Conference server.

A.   Software Asterisk PBX

Software Asterisk PBX was installed to PC Lenovo G500 with OS Windows 10, dual-core Intel Core i3 and RAM 4 GB. A freely available application *Oracle VM VirtualBox* had to be installed to the PC to ensure the required compatibility with the Linux operating system. After installation and running application *Oracle VM VirtualBox* a virtual system had to be created, named and its type and version of OS was selected. In the next step it was necessary to assign the appropriate RAM allocated from PC`s memory for the virtual system. Recommended value is 512 MB but considering a subsequent installation of Asterisk PBX, it was appropriate to assign memory with value at least 1GB. It is possible to change this setting additionally too.

When creating a virtual system, it was necessary to create a hard drive, choose the format and selection of dynamic storage allocation size. In this case, the dynamic allocation means increasing the storage of the system as needed – up to the maximum that was set while setting up a virtual system. At this stage it was important to check the storage space provided by the PC itself and set a limit of the maximum storage to a lower value in order to prevent overflow of the disk. Subsequently it was possible to apply Asterisk platform in settings of the virtual system storage by opening a freely available file AsteriskNOW.iso. Some important settings were created during the installation, such as selecting a network interface of network access of Asterisk, specification of TCP / IP settings and authentication credentials settings. Network interface can be changed even after the successful installation of the PBX – Network interface type Ethernet was used in the

design, because this type of interface forms enter to a telecommunication PLC network.

Setting the properties of network adapter for Virtual Box in PC Lenovo was necessary to ensure compatibility with Asterisk with the network. In network settings it was also necessary to allow connection for other users and to set TCP/IPv4 IP address from required network.

The PBX on AsteriskNOW platform had began to run by activating a virtual system with Asterisk installed. Then the authentication based on authentication credential set during the installation was needed. The PBX can be then configured in two ways – through the command-line commands supported by the Linux operating system directly on the virtual system interface or through a Web Interface called FreePBX. At this stage, the software Asterisk PBX was ready to manage and further configuration.

B.   End Devices of the Telecommunication Network

In PLC telecommunication network with Corinex HD-200-W2HF modems were used hardware telephones as well as software telephones. Using software telephones, it was necessary to install one of freely available software to appropriate devices. The X-Lite software telephone was selected ad installed. Its user interface is shown in fig. 3 left. Software telephone X-Lite allows the use of different codecs in support of voice and video in real time, such as G.711, G.722, or H.263. Software telephone X-Lite is easy to use and comfortable to use in enterprise environment.

Figure 3. User interface of software telephone X-Lite and IP telephone Gigaset C470I.

Software Asterisk PBX provides high compatibility with most IP phones, regardless of brand or type. Hardware telephones used in the telecommunication network design were IP DECT telephones from Gigaset C470IP shown in Fig. 3 right. The base station of telephone is powered from external adapter connected to electricity network and a wireless telephone is powered by rechargeable batteries. The telephone allows the usage of multiple codecs to support voice transmission in real time, such as G.711, G.729 and G.723, but does not allow transmission of the video.

C.   End Devices of the Telecommunication Network

Enterprise telecommunication network design was built in ground floor building with area of approximately 300 m^2. Power lines that created a communication medium had a total length of approximately 25 m. The possible placement of network components and the setting up of the VoIP

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telecommunications network based on PLC technology using Corinex HD-200-W2HF modems is shown in Fig. 4.

After the establishment of the physical communication network it was necessary to ensure logical connection between the components. The logical scheme of the telecommunication network is shown in Fig 5. Usage of static IP addresses was chosen due to the further testing network and the need for sufficient brief maintenance. IP address range was selected by network 10.0.0.0/24 – IP addresses from the range were assigned to single network components – PLC modems, PCs, router interfaces, hardware IP telephones and software Asterisk PBX.
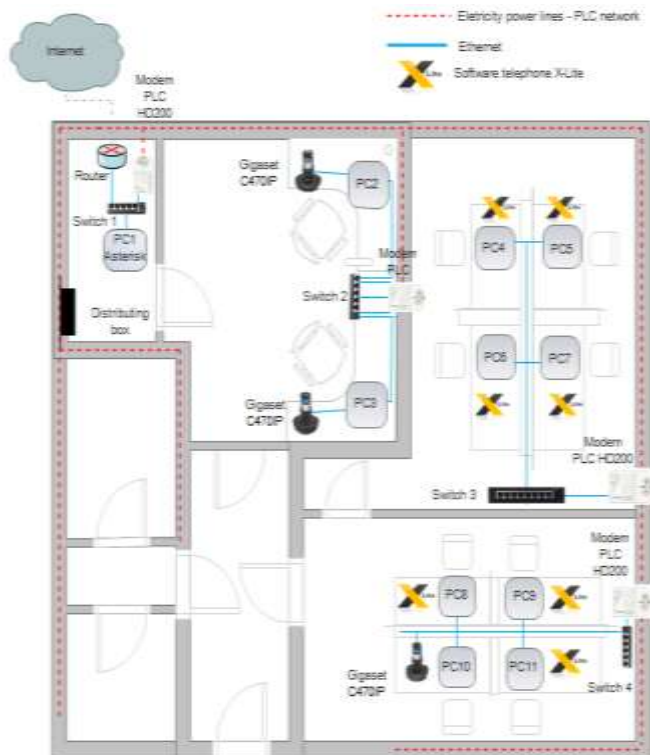

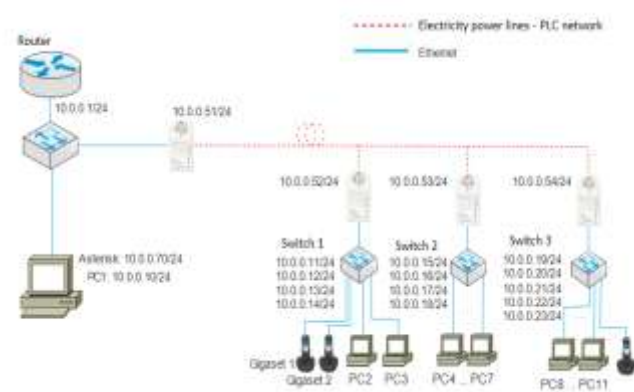Figure 4. Telecommunication network PLC HD-200-W2HF.

Figure 5. Logical scheme of the design.

Forming a local area network was implemented by configuration of IP addresses according to the scheme in Fig. 5. In the case of PCs, the IP addresses were statically set to their network adapters in network settings offered by the Windows operating system. IP addresses were also statically configured to PLC modems in the user interfaces.

D.   End Devices of the Telecommunication Network

In the next step it was necessary to draw attention to software Asterisk PBX itself which was in previous procedure successfully installed. In terms of telecommunications network it was necessary to properly configure Asterisk PBX as well as hardware telephones and to create the appropriate extensions for individual terminals. The way of assigning IP addresses to Asterisk PBX was implemented dynamically via DHCP (Dynamic Host Configuration Protocol) after installing the software PBX. Considering the way of assigning IP addresses in the design, it was necessary to change dynamic assignment of IP addresses to static as well as define the static IP address according to a logical scheme shown in Fig. 5. Network setting in Asterisk PBX can be configured using the command line of Linux operating system as follows:

[root@localhost ~]# nano /etc/sysconfig/network-scripts/ifcfg-eth0

Inserting the command it was possible to access a file of PBX ifcfg-eth0, which is used to configure network settings of the PBX. Configuration of network settings of PBX is shown in Fig. 6.

Figure 6. Configuration of network settings of software Asterisk PBX

In the file it was necessary to set these parameters:
BOOTPROTO – way of assignment of PBX IP address (dynamic/static)
NETMASK – network mask
IPADDR – needed if the static way of assignment is selected
GATEWAY – needed if the connection is above the LAN
NETWORK – definition of network of PBX placement

Almost each network component was able to provide a communication with each other at this stage of forming a local area network. Command "ping ip_address" was used to verify the connection between the components. The cross-connection between all devices including software Asterisk PBX was successfully established.

Consequently, it was possible to configure the extensions, which were assigned to individual desks. Numbering plan was formed by a set of 8xx, including numbers from 800 to 899. Configuring of extensions was implemented in the user interface *FreePBX,* in *Extensions* item of *Application* section. Creating an extension is shown in fig. 7 by inserting three necessary parameters.

Figure 7. Creating an Extension – FreePBX.

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User Extension field represents a telephone number associated with the extension. *Display Name* field is displayed on the screen of the called user and the *Secret* field represents a password for the account that is used for user authentication when registering.

Individual extension registration from the telephones required a manual setting of registration credentials to the telephones. For software telephones registration was made by setting up an account in software X-Lite. Hardware IP telephones have a user interface where it is possible to set the registration credentials. The registration of telephones to the PBX requires three key parameters – identification number, which is an extension number, IP of the PBX and the password that was set when creating the extension on the PBX. At this stage the IP addresses were set to the hardware telephones as well. All telephones were successfully registered to the PBX by correct setting of registration credentials. Extensions can be added to PBX using its console too. Entering the console is provided by command „asterisk -r". The command „core set

verbose value_0-10" was used to observe all the processes in PBX. Value from 0 to 10 means the amount of information to be displayed in output. The successful registration of extension 823 is shown in Fig. 8.

Figure 8. Registration of telephone with extension 823.

In those created and configured telecommunication network it was possible to make a phone call. The call was captured in the program Wireshark, which allows further analysis, to verify the process of creating and processing of the call between the PBX and the created extension. The progress of SIP signaling messages and the representation of the flow of RTP packets is shown in Fig. 9.
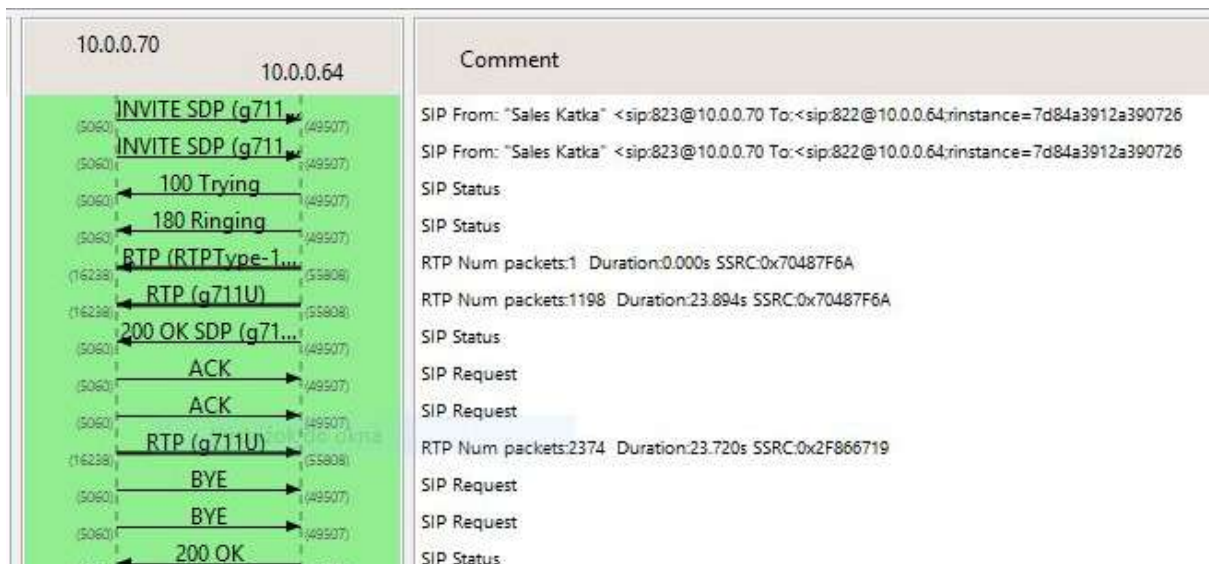

Figure 9. Creating and processing of call in the designated network – program Wireshark.

III. PREPARE YOUR PAPER BEFORE STYLING

The second segment of the enterprise network was based on the platform of PLC technology too. The newer ZyXEL PLS5206 PLC modems, with a higher transmission speed of 1 Gbps, were used in this segment of the network (Fig. 10).

Figure 10. Modem ZyXEL PLS5206.

ZyXEL PLS5206 PLC modem is based on standard HomePlug AV2 and it is also compatible with previous standards AV. It provides a theoretical transmission speed up

to 1 Gbps, operates in the frequency range 2-86 MHz and has the function of ensuring QoS, which is important especially for VoIP voice services in the designated network. These newer modems are more user-friendly in comparison with older modems. Modems must be connected to the electricity network and to the devices. Then the communication between the devices is established. The management using a Web Interface is not needed, as modems are automatically paired and the secure communications is also automatically established using AES encryption with a key length of 128 bits.

Figure 11. PBX Panasonic KX-NS500.

The hardware Panasonic PBX KX-NS500 with extension module KX-NS520 was used in second segment of the

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enterprise network. Panasonic PBX KX-NS500 shown in Fig. 11 is an intelligent hybrid IP communication system designed for small and medium-sized companies. It is easily configurable and expandable according to the needs of the company. It allows you to connect various types of phone – analogue, digital phones, SIP software and IP telephones. The advantage from an economic point of view is the possibility of using analogue and digital telephones which lead to reuse of an older communication system of the company. The PBX provides a number of useful features that can streamline the communication processes in the enterprise, including call centers without the need to use CTI server and also provides Unified Communications function. In the context of improving the communications services through statistics and analysis of customer calls, the PBX allows recording of conversations and to backup to storage device of PBX. The PBX is managed in the Web Interface, shown in Fig 12, where you can configure and manage all the PBX settings.

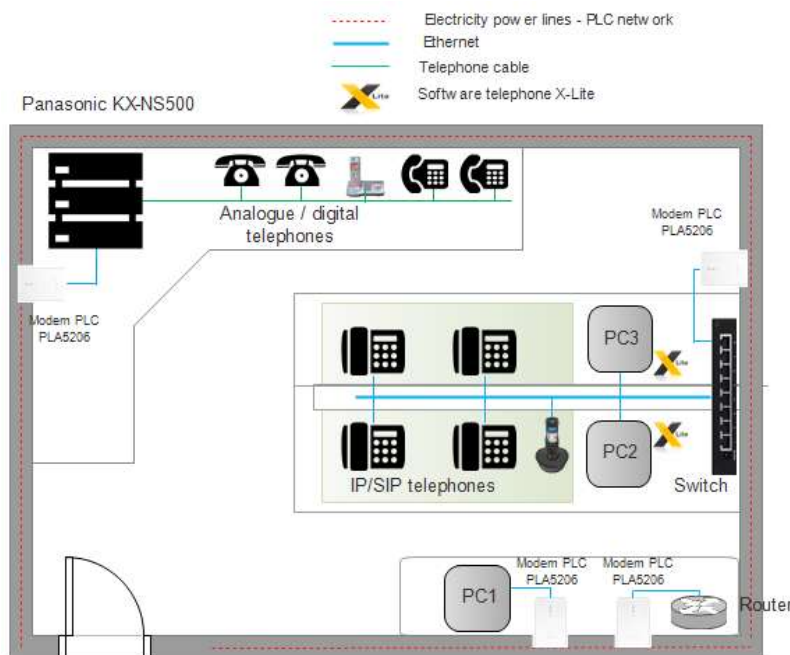Figure 12. Web Interface of PBX Panasonic KX-NS500.

Figure 13. Telecommunication network PLC PLA5206.

Similar to the previous draft of telecommunications network served by Asterisk PBX, the software telephones X-Lite were used in the PLC telecommunications network segment with PLC modems ZyXEL PLS5206 too. Hardware telephones used in this segment are listed in the Tab. I – hardware PBX Panasonic KX-NS500 provides high compatibility with most IP phones, regardless of their brand or type.

TABLE I.   Hardware telephones of telecommunication network design

IP telephones	SIP telephones	Digital telephones
Panasonic KX-NT556	Panasonic KX-UT133	Panasonic KX-DT521
Panasonic KX-NT560	Panasonic KX-UT123	Panasonic KX-DT546
Panasonic KX-TGP600		Panasonic KX-TG7120

One pair of analogue telephones and various kinds of telephones from Panasonic (Tab. I) were used in this segment of enterprise network. Compatibility between different types of hardware terminals is provided by Panasonic PBX KX-NS500, which is a great advantage because it allows the use of older telephone models not supporting VoIP communications.

The represented segment of the enterprise network with a Panasonic PBX KX-NS500 was tested in the school laboratory in the university in the configuration shown in Fig. 13.

The network was prepared analogously to the first segment. Different was just a use of end devices as well as the use of the switch. End devices were powered by PoE technology implemented in the switch. Panasonic PBX KX-NS500 was connected via the first modem to the electricity network. Second modem connected to electricity network was intended to connect the PC1. The third modem was connected to the switch with two computers PC2, PC3, as well as IP and SIP telephones. The last modem was a modem for connection to the network edge router in order to further interconnection of the network.
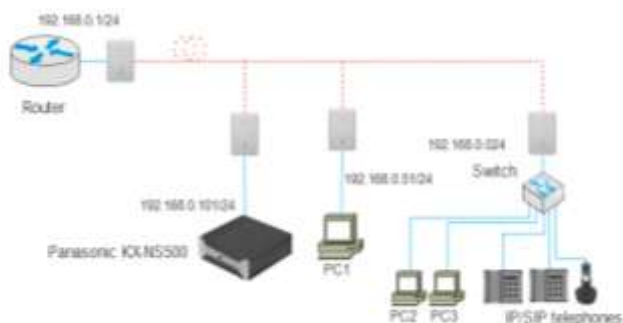

Figure 14. Logical scheme of the design.

Logical communication scheme is shown in fig. 14. The IP addresses range of the network was 192.168.0.0/24 and the PBX was set using the IP address 192.168.0.101/24. Proprietary telephones were able to register to PBX automatically, therefore the dynamic IP addresses assignment by the PBX was enable. Other components – PC1, PC2 and PC3 were manually assigned a static IP address from the range. Unlike the previous network segment, the IP address of PLC modems did not have to be assigned because their pairing and connecting to the network was fully automatic. Created logical link was verified again by command "ping IP_address".

As The hardware Panasonic PBX KX-NS500 with extension module KX-NS520 was used in second segment of the enterprise network. Panasonic PBX KX-NS500 shown in

Fig. 11 is an intelligent hybrid IP communication system designed for small and medium-sized companies. It is easily configurable and expandable according to the needs of the company. It allows you to connect various types of phone – analogue, digital phones, SIP software and IP telephones. The advantage from an economic point of view is the possibility of using analogue and digital telephones which lead to reuse of an older communication system of the company. The PBX provides a number of useful features that can streamline the communication processes in the enterprise, including call centers without the need to use CTI server and also provides Unified Communications function. In the context of improving the communications services through statistics and analysis of customer calls, the PBX allows recording of conversations and to backup to storage device of PBX. The PBX is managed in the Web Interface, shown in Fig 12, where you can configure and manage all the PBX settings already mentioned, the Panasonic PBX KX-NS500 allows automatic registration of IP phones when using proprietary telephones. In the presented network design two software telephones were used. As in the first segment of the network, it was necessary to manually create the extensions. Accounts for software telephones were created in a virtual slot in the "*Configuration*", namely in the card slot V-SIPEXT32. Firstly, it was necessary to put the card into the state OUS (Out of Service) and then the required parameters could be set – *Password* and *Extension Number*. Then the card was put into the state INS (In Service) as is shown in Fig. 15, and the account on the software telephone was successfully registered.

No.	Shelf	Slot	Port	Extension Number	Password	Connection	Current IP Address
	ALL					ALL	
1	Virtual	6	1	181	ahoj181	INS	192.168.0.120
2	Virtual	6	2	182	ahoj182	Fault	0.0.0.0
3	Virtual	6	3	183		Fault	0.0.0.0
4	Virtual	6	4	184	ahoj184	Fault	0.0.0.0
5	Virtual	6	5	185	ahoj185	Fault	0.0.0.0

Figure 15. Successful extension registration.

IV. INTERCONNECTION OF SOFTWARE ASTRISK PBX AND HARDWARE PANASONIC PBX KX/NS DESIGN

In the context of a telecommunications infrastructure of the company it was necessary to solve the interconnection between software Asterisk PBX and Panasonic PBX KX-NS500. Each of PBXs provides services to its locality – the segment of enterprise telecommunication network designed and built on PLC technology in both cases.

The designed scheme of interconnection between two segments of enterprise network, the first segment was operated by Asterisk PBX and the second by hardware Panasonic PBX KX-NS500, is shown in Fig. 16.

Introduced interconnection design of two network segments via MPLS network was not implemented due to economical terms. The required expensive hardware thus was not available. The MPLS networks are generally part of backbone networks of major providers. However, if the design would be implemented through an providers network, this method of interconnection remains to consider because it is a very reliable, fast and safe alternative. MPLS technology provides interconnection of geographically remote networks. The MPLS network is formed from a set of switches or routers that switch tagged packets. The packets were tagged at the entrance to the

network. As a result, the transfer speed of the packets is a much higher because the packets are not routed – they are simply switched. Whereas the VoIP traffic is transmit in the design, the reliable, fast and QoS providing connection is required. The MPLS network dispose of all these requirements – except the

speed, it is also very stable due to simple switching logic. It also allows assigning different levels of priority to each packet which is related with QoS guarantee. All these assumptions led to use the MPLS network in the design.
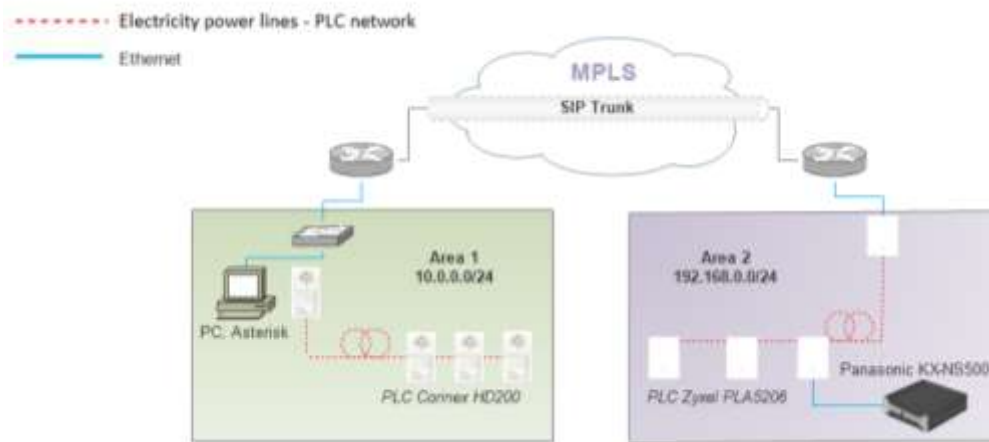

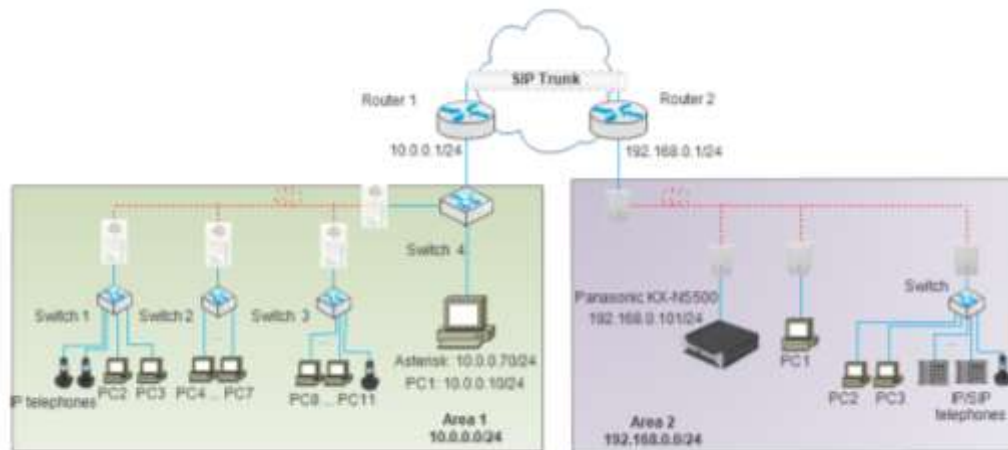Figure 16. Interconnection of networks via MPLS.

Figure 17. Interconnection between localities via SIP Trunk.

Finally, the connection between two localities was implemented via SIP Trunk. The connection was established between the two routers located on the edge of the two networks, as shown in Fig. 17. Two Cisco routers ISR 1841 were used in configuration that allowed logical connection of two network segments. The private IP address range in the first locality was 10.0.0.0/24 and in the second locality it was 192.168.0.0/24. Network between the routers represented a space for an eventual connection via the MPLS network, as mentioned in the previous section.

Many unsuccessful attempts were tried before the successful logical SIP trunk between the software Asterisk PBX and hardware Panasonic PBX KX-NS500 was established. Finally, the correct combination of configuration and settings on both PBXs sites was found which led to the successful establishment of the connection described in the following sections.

TABLE II.   SETTINGS OF THE SIP TRUNK – ASTERISK PBX

Field	Value	Short description
Trunk Name	PANASONIC3	Name of the trunk
Outbound Caller ID	182	Presentation number in outgoing calls via trunk
Maximum Channels	10	The maximum number of simultaneous calls over the trunk
Outgoing Settings PEER Details	username=182 type=peer secret=ahoj182 host=192.168.0.101	Setting up a phone number, password and IP address to be used for registration of the PBX registered to the trunk
Incoming Settings USER Details	context=from-trunk username=182 type=peer secret=ahoj182 host=192.168.0.101	
Register String	182:ahoj182@192.168.0.101/182	Registration string

The settings and configuration of SIP Trunk were carried out in the Web Interfaces of both PBXs. It was necessary to set a login number, password and IP address of opposite PBX to create a SIP trunk. The configuration and settings of the SIP Trunk were implemented for Asterisk PBX in the "*Trunks*" section of the "*Connectivity*" item. The values of appropriate fields were set according to a Tab. II. It was also necessary to set value 5060 to port in „*Asterisk SIP Settings*" section of the „*Application*" item.

The configuration and settings of the SIP Trunk were implemented for Panasonic PBX in the virtual slot "*Extension*" of the "*Configuration*" section. The "*Port Property*" item had to be put into an OUS state as is shown in an extract of configuration table in Web Interface. The username, secret and IP address of Asterisk PBX were set up in the appropriate fields in the section *Port Property - Virtual SIP Extension*.

No.	Shelf	Slot	Port	Extens. Numi	Password	Connection	Current IP Address	
	ALL ▼					ALL ▼		
1	Virtual	6	1	181	ahoj181	Fault	147.175.103.178	1
2	Virtual	6	2	182	ahoj182	INS	10.0.0.70	
3	Virtual	6	3	183		Fault	192.168.0.141	
4	Virtual	6	4	184	ahoj184	Fault	10.0.0.68	1
5	Virtual	6	5	185	ahoj185	Fault	0.0.0.0	1

Figure 18. Port Property – Virtual SIP Extension.

Connection	Channel Attribute	SIP Server Name (100 characters)	SIP Server IP Address	SIP Server IP Address for Failover	SIP Server Port Number	Authentication ID (64 characters)	Authentication Password (32 characters)
ALL ▼	ALL ▼						
INS	Basic channel		10.0.0.70		5060	807	jano123
Fault	Basic channel		192.168.0.2		5060	805	jano123
Fault	Basic channel		192.168.0.2		5060	123456789	123456789
Fault	Not Used				5060		

Figure 19. Port Property – SIP Trunk.

The virtual card was put into a state INS again and the required status INS appeared also in *Connection* column as shown in fig. 18. Then the rules for incoming and outgoing calls were defined in both PBXs. The Panasonic PBX has outgoing calls management in the "*Numbering Plan"* of *"PBX Configuration"* section. Here is the area code for dialing via various slots selected. In the connection via virtual slot was selected area code of number "9". The rules management for incoming and outgoing calls of Asterisk PBX is in „*Inbound Routes*" and „*Outbound Routes*" items of "*Connectivity*" section.

In further it was necessary to define appropriate area codes on Asterisk PBX site. The area code for calls via SIP Trunk was selected as the number "1" and the rule was associated with the SIP Trunk called PANASONIC3 that has been prepared according to the Tab. II. The end users from the area 1 thus dial the telephone numbers to area 2 with the prefix "1". In reverse process, the users from area 2 dial telephone numbers of end users from area 1 with the prefix "9".

It was necessary to set the virtual card to a proper INS state and the expected status INS displayed in a column *Connection* (Fig. 18) – the connection was successfully established. Failed or unsuccessful connections were represented by Fault state. In this case the successful SIP Trunk connection was established and the calls via SIP Trunk were enabled by the rules of incoming and outgoing calls definition on both exchanges. This procedure is described in the next section. Then it was possible to make calls via the SIP trunk from network segment in the area 2 with the Panasonic PBX KX-NS500 to network segment in area 1 with Asterisk PBX. The SIP trunk connection described in the above procedure was implemented also in a second process (described in the next section), which allows the calls from Asterisk PBX to Panasonic PBX by the appropriate configuration and options of both PBXs.

In the further procedure it was necessary to set extension by telephone number 807 and an appropriate password "jano123"definition. The aforementioned was set in the Web Interface of Asterisk PBX, "*Extensions*" section of "*Applications*" item as is shown in fig. 19. The configuration of the SIP Trunk in the Web Interface Panasonic PBX KX-NS500 was executed in a virtual slot "*Trunk*" of "*Configuration*" section. First, it was necessary to set the "*Trunk Card Slot*" itself, which can be configured only in OUS state. The port number 5060 was set in the "*Port Property*" section and then the SIP trunk could be set as shown in Fig. 18.

The connection described above was functional, tested and ready to implement two LANs separation as is shown in fig. 16. First of all it was necessary to prepare the configuration of Cisco routers. It should be emphasized that the routers in the topology represent an ability to become routers at the edge of the MPLS network of provider such is shown in Fig. 16. After verification of the connections between the networks by the "ping ip_address" commands, the IP address of the Asterisk PBX in its network configuration to IP address 10.0.0.7/24. Also the default gateway was set – the IP address of the network interface of the ROUTER 1: 10.0.0.1/24. The IP address of Panasonic PBX KX-NS500 could be maintained but the default gateway had to be set in network settings of the PBX – IP address of the ROUTER 2: 192.168.0.1/24. Then the connection between two PBXs was verified – see in Fig. 20 and 21.

Figure 20. Connection verification – Asterisk → Panasonic.

In the next step it was also necessary to change IP address of Asterisk PBX in SIP Trunk configuration and after restarting both PBXs the SIP Trunk was verified by processing calls between two areas. The call was captured by Wireshark to show the procedure of establishing and processing of the call (see Fig. 22).

Figure 21. Connection verification – Panasonic → Asterisk.
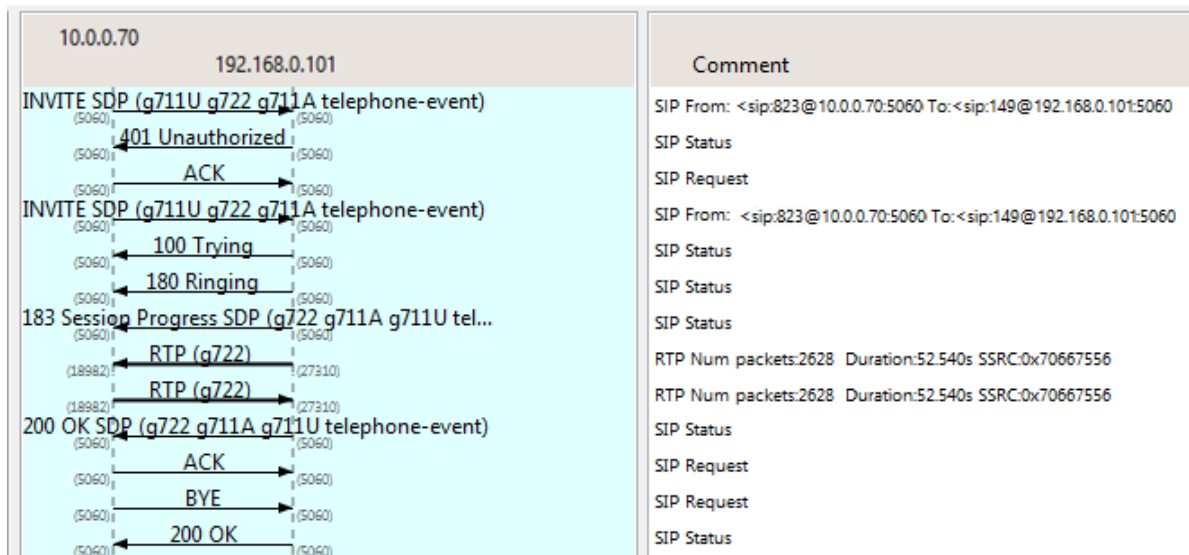

Figure 22. Call processing via SIP Trunk.

The interconnection of multiple PBXs in one company is one of possible solutions of private telecommunication network. Many companies prefer to service its telecommunication network with one central PBX even if the offices are in different cities. However, creation of decentralized solution for interconnecting PBXs is mainly used in the case of multinational companies that have their offices in different countries or continents. Then it is much more reliable and safer to establish interconnection of several PBXs, each operating in appropriate office. In terms of redundancy it is advantageous to have several interconnected exchanges for such multinational companies with offices worldwide. If there is a fault in one of the PBXs or one of the connections was disturbed, the other PBXs can take over its function and the failure will not interrupt the communication in the company. Another advantage is the load balancing of traffic between exchanges - traffic is distributed between the PBXs, thus achieving their optimal use. If the PBXs are interconnected in practice, quite often the two existing systems in two different locations are interconnected what significantly saves

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communications costs. Calls between sites are not directed to the public telecommunications network, but directly processing between exchanges. So, one of the major reasons for the connection of the PBXs is definitely cost saving for the company's communications.

V.   CONCLUSION

The purpose of the article was the telecommunication infrastructure design based on backbone PLC technology, composed of two different telecommunication systems – software Asterisk PBX and IP Panasonic PBX KX-NS500. Fundamental part of the design was to create a connection between software Asterisk PBX and IP Panasonic PBX KX-NS500. The first part of design led to form the Asterisk PBX network segment and to installation of Asterisk PBX itself. In the second part the realization was based on the hardware Panasonic PBX KX-NS500. Both segments were based on PLC technology. The connection between two segments was implemented via SIP Trunk using two edge routers. This solution enables perspective connection via MPLS network of provider. While implementing the connection, many concepts of possible solution were tried. It was a result of series unsuccessful attempts and tries of different combinations of settings and PBX`s configurations to find a successful connection because of the combination of two different platforms of PBXs. Finally, the correct combination was founded which led to the successful connection.

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