VoIP Technology as an Example of Integration of Business Services on the Example of Philips' European Service Center in Lodz.

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Abstract - **The article presents the results of work relating to the use and functionality of VoIP telephony and business applications at Philips ESC in Lodz with NEC MA4000 application.**

I. Introduction

Development of the Internet today has brought many new opportunities. One of the areas, which is extremely important for any kind of business is telephony Voice over Internet Protocol (VoIP), a voice message by the network packet with IP protocol. Due to the popularity of VoIP gained can be called one of the most important ICT services. Is a technology that affects all users of telephony, Internet, fax, email and network. Main customers of this technology are primarily business and enterprise. The implementation of VoIP systems makes it possible to reduce the cost of telephone calls. VoIP technology uses multiple disciplines of communications technology. Its aim is to revolutionize the way you communicate primarily by increasing the speed, reliability and availability of the Internet. Under her influence is also Internet Protocol telephony, digital, analog, and many other networking technologies.

VoIP technology is future-oriented, but still must overcome many problems that are associated with compatibility and standardization.

II. Operation VoIP

The IP technology can emerge three types of interfaces associated with VoIP:

- IP telephony in the enterprise,
- Comprehensive VoIP solution for corporations,
- Internet telephony for private customers.

All three applications make use of signal transport technology and operating principles based on VoIP technology. What connects these audiences is the fact that in each of the three cases used hybrid solutions (IP and TDM).

For the first group of the most important are:

reliability, safety of the talks, high quality conversations, flexibility allows the use of additional equipment and raise the additional features: voice mail, conferencing, recording the talks, and more.

Private person using Internet telephony poses mainly to the low cost of the call, and wide availability.

While the classification of Internet telephony solutions (Figure 2.1), consider three aspects. First, we use signaling protocols, and secondly how to choose the communication system infrastructure and what we are interested in network coverage.

Acceptability of voice quality for customers is the essence of the use of VoIP technology. A condition that must be met in order / in fact had a raison d'etre is to delay the voice signal to less than 150ms (Figure 2.1). What's more - absolutely voice packets must be delivered to the recipient in the correct order, and frequent the same time differences. All this depends on the audio coding method, the signal transmission time (directly related to the delay of packets) and interference (loss of data frames). Operators must develop monitoring systems to ensure the quality of the conversation. Moreover, the additional condition is prioritization and buffering voice packets in IP network nodes.

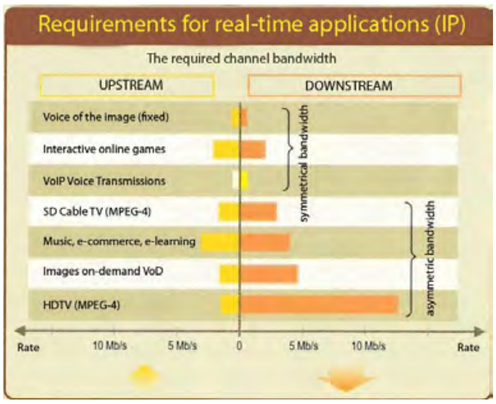

Figure 2.1 Requirements for real-time applications (IP)[1]

There are two standards, which include the older H.323 / PSTN and more modern SIP / PSTN. This is an important point of the VoIP technology that allows communication between packet network and public network PSTN. It is possible to exchange information between all kinds of communication technologies such as telephony, analogue, digital, IP, Videophone, computers. Legitimacy of goals is justified in situations when clients try to connect to a network outside of the packet. Gate sets the highest priority for IP telephony, eg if the data transfer receives PRIORITY lower. It is directly related to the requirements of obtaining real-time during a call. Signaling protocols are designed to manage the system and directing them across the network. Applications that use these protocols must also keep refresh the information on the availability of bandwidth. This creates a contact between the users of all types of networks.

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Network architecture from the viewpoint of the H.323 form elements such as:

●Terminals have a full opportunity to make a call, send and receive on a full-duplex data stream. Terminal device is both dedicated to making calls,

●Guard (called gatekeeper) - is responsible for the management of terminals, gateways and MCU server,

●Gateway - This is a part of the voice infrastructure, which is responsible for IP calls to other types of networks and for establishing and disconnecting connections in networks that are linked together,

●Server MCU (Multipoint Control Unit) - it serves to separate streams, and to transmit them to the terminals. The server consists of two elements:

●Multipoint Controller (MC) - is responsible for the exchange of information,

● Multipoint Processors (MP) - is responsible for handling the data stream.

In the case of a VoIP network based on SIP logical entities are:

●User Agent (UA-User Agent) - aims to initiate and terminate sessions by exchanging messages also with another entity

●Proxy Server - is a mediator because it acts as a client and server. The server is designed to accept and fulfill requests that are sent by other clients

●Registrat - this is a server that accepts requests a Register. It aims to update the database that contains information to establish the connection.

●Server redirection (Redirect Server) - It accepts SIP requests and maps the SIP address of the client paying them. This server does not forward requests to other servers.

Business IP telephony solutions, the nature of the standards used to create a VoIP phone system with higher-level functions in addition to providing traditional voice features in LAN, telephony refers to the voice of transportation systems using generic procedures for signaling (eg SIP, H.323).

The main objective of VoIP is merely a reduction in external costs, phone calls and gradually accustomed users to a new form of VoIP packet handling.

III. MA4000 APPLICATION MANAGER

Software that has been used to manage the VoIP services provided by NEC MA4000 Manager version.

Home application is intended to inform the user about the tasks currently performed and the possible alarms and system error. MA4000 System is a centralized management system for web servers communicate.

Central MA4000 management system is designed to improve the management of subscribers, the load control of network resources and full control of call costs.

MA4000 is a safe, easy to use, efficient and reliable tool for improving management of the telecommunications system. Additionally, the system offers advanced alarm handling and full control of call costs.

MA4000 application is divided into five categories, which include:

●Administration panel

●System settings, allows to manage voice systems.

●Management Panel Users / User groups and devices

IV. CONCLUSIONS

This article introduces the principle of operation, a wide variety of VoIP applications, yet still young and rapidly growing field of computer science.

By using VoIP technology, user is able to use a single phone and is always available at a single telephone number.

VoIP is beginning to be seen as an alternative for local calls, where a network packet can be much easier to implement services such as teleconferencing.

MA4000 system installation in an international company, despite the very high cost of purchase, was a very good choice. In use the system, it turned out that only the external conditions such as lightning are able to disrupt the system, through loss of electricity supply. With the result that there has been damage to one of the servers. The only solution was to exchange server. All the settings that were stored on memory cards, which remained preserved. Ratio of memory cards to the new server allowing the rapid restoration of the whole system without the need for reconfiguration.

The problem that occurred when using the system, it was that when using the phone recognizes the phenomenon of one-way audibility. The cause was a configuration error on Business Connect application, and exactly the wrong time for setting session summary IVR connections, which proved to be too short. Increasing this parameter in the application settings solved the problem.

For the business needs of several groups established to implement the call in a smooth, thanks to the perception of customers regarding their service was consistent, as evidenced by the immediate settlement of the perception of the company, which according to them, and provided a very professional level of service.

It can be concluded that the most important condition for the development of VoIP technology is an economic bill, which must be beneficial for companies. If the price of the equipment and the creation of networks and connection to the central public will continue to decline and in the end will be lower than the same panel-based network such as ISDN, the technology will be able to replace POTS and ISDN services. This technology brings great benefits when conducting such long distance calls, international.

V. BIBLIOGRAPHY

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