Implementation of Cost-effective VoIP Network

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***Abstract*—This paper describes a sample test implementation of an cost effective Voice over IP network. The implementation is currently taking place at Department of Microelectronics and Computer Science in Technical University of Lodz, Poland**

***Index Terms*—VoIP, Voice over IP, SIP, Asterisk**

I. Introduction

Voice over IP technology is considered to be a feature-rich replacement for traditional POTS[1] networks. During recent years multiple manufacturers have decided to develop VoIP hardware equipment, such as IP Phones, Analog Telephone Adapters (ATA), or media gateways that can be used in developing Voice over IP network in enterprise conditions. Real time transmission of voice data over the IP network can have significant benefits over traditional telephony networks such as:

- Reuse of an existing computer network for the telephony network which can minimize costs introduced by maintaining and improving two wired networks at the same time.
- Ability to connect many different phones to one network wire, by using inexpensive hardware such as SOHO class switches.
- Ability to reach to places with no wire infrastructure by using the wireless network
- Ability to include modern mobile phones supplied with VoIP software in the telephone network, allowing them to use every feature deployed on the VoIP PBX.

When choosing a solution to implement a Voice over IP network often the cost of the hardware is one of the most important factors. As VoIP terminal devices are becoming cheaper, the equipment needed to interconnect digital or analog telephony network with VoIP, as well as the central server can be among the most expensive parts of the whole system. When dealing with the heterogeneous network there are however, software solutions which can replace hardware VoIP PBX. The use of the software solution can benefit from reduced Total Cost of Ownership (TCO)[5] as well as the ability design VoIP services more specific to the given telephony network.

1 POTS (Plan Old Telephone Service) – a voice service provided to home or small office. Usually consisting of analog phone lines.

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II. Project criteria

The aim of presented assignment was to implement VoIP technology in the building of Department of Microelectronic and Computer Science (DMCS) at the Technical University of Łódz. Practical implementation gives opportunity to discover and face issues, which are hard or in many cases impossible to foresee when considering theoretical knowledge.

Decision about migration on VoIP was driven by following factors:

- Cost-savings. Since DMCS cooperates with numbers of organizations and academia from all round the world, appliance of VoIP technology is the best possibility to significantly cut the toll charges.
- Mobility. Many Ph.D. students and academic teachers, carries their researches in other countries (Germany, France etc.). With VoIP it is possible for them to be reachable, all over the world under the same telephone number. Caller does not need to know, where the called person currently is.
- Functionality improvement. Appliance of VoIP bring wide range of new features including voice mail, Do not Disturb, advanced billing control, full control of users privileges, intercom etc.
- Support of a new technology. As scientific unit and department of computer science, DMCS is also supposed to propagate new technologies and solutions. Implementation of VoIP infrastructure gives future students opportunity to investigate and get familiar with VoIP technology, which is expected to influence and change the telecommunications world.

Additionally, present feature set should be extended, providing new functions mentioned above and demonstrate them to the users, so that they will be aware of them and benefit from them

III. Basic Terms

Before discussing details of introduced assignment it is important to have basic knowledge about VoIP technology. Telecommunication is most probably the last major electronic branch of industry, which until now remained unchanged over many years, basing commonly on inventions from last century. Together with the improvement of digital networks, emerged possibility to offer totally new standard for telecommunication – Voice over IP. The real value of VoIP is that it allows voice to become nothing more than another application in the computer network[6]. That gives myriad of new possibilities.

VoIP allows to achieve significant savings not only in calls costs but also in the network maintenance, since data and voice can be handled on the same type of the network. VoIP offers countless services that older telephony systems cannot provide. Offered flexibility can be achieved, because VoIP is designed to interface with an IP network, following its rules, utilizing its protocols and as a result accessing numerous applications that exist in the network. IP standard is widely known, therefore it is comparatively easy to develop new applications and new features obeying IP Protocol rules instead of writing vendors specific application. In this environment new voice applications are developed, similarly to common data applications written in Java or C, which results in deployment time reduction from years to weeks or even days. VoIP telephony services are interoperable, meaning they work on all kinds of networks using IP protocol. VoIP services, will work with any IP-enabled device including IP phones, computers, modern GSM phones or Personal Data Assistants (PDAs).

On the other side VoIP struggles with some issues, which need to be taken into account when considering VoIP implementation. From the inception of VoIP, many improvements have been required for the network and communications models, in order to decrease and eliminate negative effects of voice transportation over the Internet. Traditional telephony has been developed over long years and solely with the telephone conversation in the background. Over these years, hundreds of signalling protocols and communications protocols were designed and introduced to many types of telecommunication networks such as PSTN, X.25, ISDN (Integrated Services Digital network), SMDS (Switched Multimegabit Data Services), frame relay or ATM (Asynchronous Transfer Mode). Internet was even not expected in the development phase to be used to carry voice data. Conditions which needs to be met transporting raw data differ entirely from those for voice.

Difference results mainly from the human sound perception. All voice packets need to be reassembled in correct order, without errors and this task has to be finished within less than 150 milliseconds[1]. Longer delivery times or packets getting lost cause degradation of perceived conversation quality and can lead to difficulties in carrying on a conversation. Due to the layered architecture it was possible to design desired protocols only at the higher layers of TCP/IP model, without demand to reorganize bottom layers. Mechanism for carrying on conversation over the internet includes signalling transactions and exchange of media stream in each direction. There are several protocols designed to complete this tasks and all others necessary requirements for successful VoIP conversation. Some of them deal only with signalling, others with data transport between the endpoints.

Undoubtedly SIP (Session Initiation Protocol) is the most widely deployed protocol[4]. Its success SIP owns mainly to freely available documentation and its simple premise: each end of the connection is a peer. There is no master and slave and both parties have the same rights, where protocol negotiates capabilities between them. It makes SIP a relatively simple protocol, which is furthermore based on extensively known protocols like HTTP (Hyper Text Transport Protocol) or SMTP (Simple Mail Transport Protocol). As the name implies, the primary function of SIP is session initiation. It does not mean however, that it is an exclusive task for SIP. SIP is additionally designed to perform basic functions of a communications network like address resolution, call control, Quality of Service as well as non session related functions such as mobility, message transport, event subscription, notification, authentication and extensibility. Sip does much, but in fact, it requires other protocols including above all RTP(Real Time Protocol) to enable voice communication. With presented set of protocols the next step on the way to set up the conversation over internet is PBX (Private Branch Exchange), where the internet telephones will register to.

IV. Asterisk PBX software

Asterisk[2] is an open source software package. It is provided free of charge comparing to hardware or hybrid PBX[3] systems which are notoriously expensive. In fact , the costs of PBXes were the main reason, which has made authors of Asterisk – Mark Spencer and Jim Dixon to begin this project.

Asterisk is PBX platform that runs on many operating systems including Linux, BSD, Mac and recently even Windows (Windows is however not recommended for this purpose since near real time operation is required). It is an open source toolkit for telephony application and feature-rich call-processing server. Asterisk can be standalone system or used with previously existing PBX or VoIP implementation. It can be used to manage internal VoIP calls within Local IP Network or in conjunction with specially designed hardware to interface with PSTN networks. Asterisks open source nature causes that it is constantly improved and developed. Currently Asterisks community counts thousands of participants including programmers, telecommunications professionals,

2 Asterisk PBX homepage – www.asterisk.org

3 PBX (Private Branch Exchange) – a telephone exchange system designed for home or small office use.

networking professionals and information technology professionals. Such a great support results in a rich set of features, applications and support for all technologies offered by Asterisk platform. Asterisk utilizes many protocols including H.323, SIP and IAX and can interoperate with almost all standard-based telephony equipment using relatively inexpensive hardware. At the same time, almost all leading telephony hardware manufacturers take Asterisk into account and guarantee compatibility with this software.

V. System Hardware

Once the particular PBX software has been chosen, the main issue when deploying a software PBX is choosing a suitable PC or server platform to run the software on. The performance of the platform is the key requirement for the successful completion of this assignment. Shortage of the performance can result in the quality degradation or even inability to carry on the conversation. There are many factors, which may influence overall system performance and therefore there is no simple and clear answer for the question of hardware configuration for Asterisk. Generally Asterisk as a PBX platform, is considerably efficient . It has been shown[3] that even with Pentium 100Mhz box with 64MB of RAM it was possible to build PBX for small business. Moreover, every new version of Asterisk provides performance gain. According to information from the Asterisks community [3], Asterisk in version 1.2 is able to deal with around 220 SIP calls, Asterisk 1.4 scales much better and can handle nearly double the call setups/seconds, whereas version 1.6 shows a SIP performance increase compared to 1.4 of factor 3 to 4.

A. VoIP analog gateway

When it comes to telephones, two types of them are to be used: analog phones and some additional IP sets. In order to make use of existing analog phones and enable them to place and receive VoIP calls, analog VoIP gateway must be deployed. Analog gateways allows to create cost-effective hybrid IP and take advantage of the benefits of VoIP communications while preserving investments on existing analog phones. They convert telephony traffic into IP for transmission over the data network. They are produced by several voice and video products manufacturers as external units or as PCI/PCI Express cards. For the purpose of discussed implementation Grandstream analog gateway model GXW 4024 has been chosen. Grandstream products, have good reputation in VoIP area, they offer satisfying conversation quality and price requirements. The chosen gateway provides 24 FXS analog ports, allowing to connect up to 24 phones, which is currently half of the capacity required by the developed system. Therefore 2 units of this type of hardware have to be used.

B. IP phones

Also a group of IP phones and ATAs[4] had to be chosen. After reviewing different phone capabilities the Linksys SPA IP phone series was chosen. Two different models have been used:

Linksys SPA 922 – IP phone providing monochromatic display, auto-answer function, loudspeaker, with support of the PoE[5] standard.

Linksys SPA 962 – IP phone providing up to six VoIP lines, color display, PoE support, auto answer function and loudspeaker.

The Linksys SPA IP phone series have been in use for a few years, and it is also noticeable, that its capabilities are constantly improving by the use of newer firmware revisions. One of the latest revision provided functionality to use directory servers such as OpenLDAP, in telephone number lookup. Usage of the LDAP server may facilitate integration of large numbers of phones within an organization providing one central database for use with the phone books on all phones.

C. ATA devices

To facilitate the transition between analog and IP devices, and to support non standard terminals, such as fax machines, a set of ATAs was chosen. ATAs provide the capability to connect a standard analog phone to the VoIP system. Particularly the Linksys SPA2102 ATA provided to be a reasonable solution, providing such functionalities as: connection up to two analog devices, FoIP[6] support using the T.38 protocol and rich feature set present also in the SPA IP phone products.

D. Server hardware

Performance of the server is crucial for maximum number of concurrent calls and more advanced calling features. The more concurrent calls and more complex services the more processing power is required. Number of concurrent calls is also important from scalability point of view.

The following configuration has been chosen and is expected to satisfy performance requirements:

- Processor: Intel Core2 Quad Q8200 2,33 GHZ, 4MB cache, FSB 1333Mhz
- Memory: 2x1024Mb RAM DDR2 1033Mhz, Dual Channel
- Hard Disk Drive: 250 GB Seagate , 32 MB cache, 7200 rpm, SATA II Interface
- Main board: Asus P5K SE
- Graphic card: GeForce 8500GT

Usage of an integrated graphics card was preferred, although in the given price range no suitable products were found. The server main board was chosen especially to

4 ATA (Analog Telephone Adapter) – a device that allows connection of a standard analog phone device (such as standard phone, or fax machine) to the VoIP network.

5 PoE (Power over Ethernet) – standard allowing to power network devices via an Ethernet cable connected to specific hardware (such as PoE-enabled switch)

6 FoIP (Fax over IP) – a set of specifications allowing for real time transmission of faxes through VoIP network

provide support for additional two FXO cards, which usually take two PCI slots each. There should also be a possibility to install another card later, for example a 4 line VoIP GSM card allowing to further reduce costs of the cellular calls.

VI.Implemented features

At the beginning of the VoIP implementation following set of the features, which are supposed to be available for a user, has been specified:

Voice Mail – as mentioned, feature giving possibility to leave voice message when calling party is unreachable.

Conversation recording – user should be provided with the ability to record conversation placed through PBX when necessary in order to review the stored conversation at a later time. Asterisk PBX supports various formats used for voice recordings including wav, gsm and mp3. Recordings are stored incrementally during the progress of a conversation.

Conference Calls – user should be able to set up conference call or to join and participate to the available conferences. In order to avoid unwanted user to take part in conference which are not addressed to him, it should be possible for the user initiating conference to specify PIN number, asked later on when joining the conference.

Intercom – as the name implies provides intercom functionality making use of telephone infrastructure. Intercom should only be used by the users with special privileges, given by the administrator. When used, all IP telephones (if the provide this functionality) are automatically answered and it and the caller is allowed to make an announcement.

Call forwarding when not available - feature, which enables user to set the telephone number, which calling user should be directed to when his phone is busy or unreachable.

Caller ID replacement – feature implemented in order to support the analog lines which are used in the facility. When making a call through a land line, the calling number should be replaced by the number assigned to that line. Caller ID replacement also allows an administrator to force Caller ID for a certain phone, preferably setting it to the name and surname of the owner.

Ring tones assigned to calling number – it should be possible to set particular ring tones for the particular calling number.

Additionally, given set of the features has been extended with some additional administrative permissions including configurable permission for making local / internal calls, Interstate calls and finally permission for placing the most expensive international calls. Features listed here, are only the major features, which has been developed in presented assignment, nevertheless designed system can be without difficulties extended with new features.

Listed features have been implemented in Python scripting language. Their appliance and integration with Asterisk was possible thanks to AGI (Asterisk Gateway Interface) API[7] provided with Asterisk. AGI allows to integrate the Asterisk PBX with external programs or script written in a commonly known language such as Perl, Python or JAVA. Use of the Asterisk Gateway Interface allows for performing tasks that are too complex or even impossible to develop by using standard PBX configuration. These tasks may include local database access, Web Services integration, LDAP directory lookup and more, provided they may be implemented in the used programming language.

VII.Conclusions

To conclude, requested set of features has been provided. Build solution, due to its design based on the database is very flexible and fully configurable. Every user has its own account, where it can be defined, which options should be activated and which should be prohibited. Furthermore, some additional features for better system administration have been implemented in order to give the system administrator better control over the registered users. It is possible for administrator, to lock any feature for a particular user when necessary. Presented implementation of VoIP, have shown that VoIP aside from significant cost savings, brings flexibility, broad set of features and possibility to perfectly adjust implemented platform to the business needs. It is also possible to firmly state that with correctly designed VoIP implementation, VoIP is highly reliable and offers quality comparable with PSTN.

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7 API (Application Programming Interface) – a set of programming language functions provided by the application manufacturer in order to provide an interface for using an extending an application.