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Voice over Internet Protocol

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Abstract.

Voice over IP or telephony via the Internet, characterizes a set of protocols/technologies [H.323, Session Initiation Protocol (SIP)], which provide vocal conversation in real time with relatively good quality at little or no cost, thanks to worldwide broadband connections. Traditionally, such conversations took place exclusively through a PC which was connected to the Internet and with the aid of a microphone, headphones and the appropriate software (soft phones). The call ended up in another similarly equipped PC at no extra charge other than the one needed to access the Internet, since this specific type of communication does not require a provider of standard land-line services, but only the Internet. In addition to this, there are autonomous telephone devices (VoIP SIP Phones) and analog telephone adaptors (ATA) on the market which directly connect to an IP network, like the Internet. With the right adjustments and assembly and without the need of a PC, use of this service is facilitated making VoIP even more accessible to its users.

The object of this paper is the study, realization and operation of an initially autonomous system of telephony via the Internet protocol (VoIP) and subsequently, its connection to other telephony systems (either via the Internet or traditional telephony), for their intercommunication, using the advantages of VoIP. First, the existing phone network [Public Switched Telephone Network (PSTN)] as well as its development is retraced. Then, VoIP technology is analytically presented along with its functions, protocols used, as well as its advantages and disadvantages. The quality of the service (QoS) offered by VoIP and the factors that affect its quality are also studied. As we continue, the importance of IP Private Branch eXchange (PBX) and the additional services they provide will be presented. In conclusion, an application of VoIP for military use will be executed so as to enable the military to reap the benefits it offers.

Keywords: PSTN (Public Switched Telephone Network), VoIP (Voice over Internet Protocol), SIP (Session Initiation Protocol), H.323 Protocol, QoS (Quality of Service), PBX (Private Branch eXchange), IP sec (Internet Protocol security).

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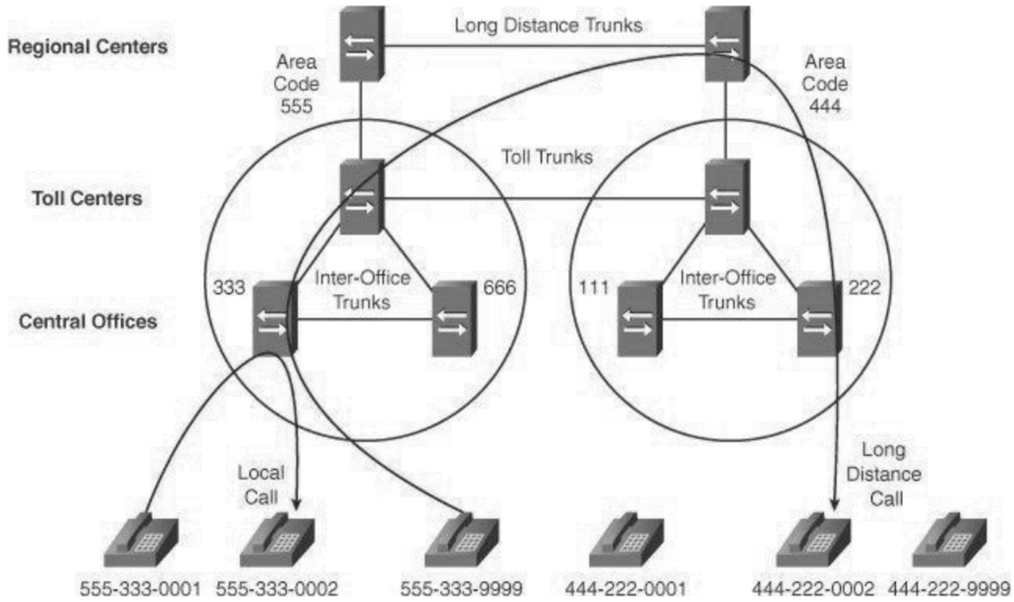
The aim of this paper is to illustrate a comprehensive overview o the interconnection of common telephony centers through the Internet, as well as to understand Voice over Internet Protocol (VoIP) technology and the importance of its application by the military. On these issues, there is already an extensive and exhaustive Bibliography. The interested reader is referred to key references [1-4], as well as the literature cited therein.

Before we start talking about broadband phone services and internet telephony, it would be useful to trace back to the past, so as to better understand the mode and the need that led to the creation of the current traditional telephony system, widely known as Public Switched Telephone Network (PSTN).

It all started when Alexander Graham Bell, who was a Scotch instructor rather than a scientist, almost accidentally discovered the first analog telephone back in 1876. That happened because Bell’s initial purpose was to send a lot of independent telegraph signals over the same circuit. Thus Bell noted that his equipment could transmit the human voice. The key of success lies in the simplicity of the whole philosophy: two wires, a handset and a microphone. That was all that somebody needed in order to communicate with somebody else, independently of the distance that separated them, at least as far as the user was concerned. Obviously, some more complex functions are needed so that communication can be achieved. This is where things become even more interesting.

Bell’s invention was the beginning of a rapid development of telephony. In 1881, the first European telephony network, which had 49 subscribers, was installed in Amsterdam. During the first years, every telephony network operated by calling the operator and ordering the connection. In specific, if someone wanted to make a call, they would lift the handset and crank the induction. In this way, a signal could reach the operator, who would convert the order into communication with a subscriber through a connection to the central telephone table.

A major problem that classic telephonic centrals had to face was the restoration of communication between distant interlocutors. Therefore, the telephony system was set up with a hierarchical structure, which can be seen in the slide.



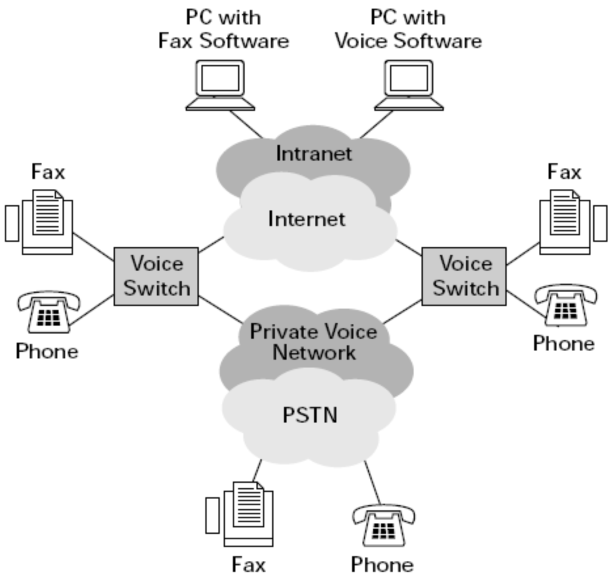
The Central Offices are the telephonic centrals that constitute the bedrock of the hierarchy; their aim is to connect the telephony lines of subscribers from their homes or workplaces with the telephonic network. The Toll Centers comprise the middle line of the hierarchy; their aim is to interconnect the local telephonic centrals of various areas, through the appropriate linkage. The Regional Centers are on the top of the hierarchical structure; their aim is to interconnect the Toll Centers of a specific geographic area. The Regional Centers usually edge into remote calls, like from one country to another.

As time went by, the increased telecommunication demands, the introduction of the Internet, and the demand for more qualitative services, led to the need to upgrade and improve the current PSTN network. This was completed in 1988, with the advent of Integrated Services Digital Network, widely known as ISDN, which provided the subscribers with more qualitative services, and they also introduced digital networks into the area of final users.

The ISDN is the evolution of the PSTN and it is constituted by a sum of telephony models that allow the simultaneous digital transmittance of picture, sound, data and other services through it. The basic characteristic and advantage of the ISDN is that it incorporates sound and data in the same line. In addition, it provides the subscriber with the opportunity of accessing the network through two connections; the Basic Rate Access (BRA), offering a bandwidth up to 128 Kbps and the Primary Rate Access (PRA), offering a bandwidth up to 2 Mbps.

As time went by, there was rapid development and spread of IP networks. This resulted in the utilization of IP networks for real-time data broadcasting telecommunication services. The main service is voice transmission. The Voice over Internet Protocol technology, also known as VoIP, allows voice transmission through IP networks, such as the Internet. The VoIP has changed (and it is still changing) the way that we communicate, by gradually replacing the traditional telephony.

Nevertheless, there are numerous ways to implement a VoIP network. It can be structured on any network which is based on an IP protocol, such as local wired or wireless networks, or even the Internet. Even current mobile telephony networks have already began using VoIP, thanks to the exploitation of the potential of “smart” devices and the respective Internet services that the providers of mobile telephony offer.



A VoIP network can also be interconnected with PSTN networks, including mobile telephony networks. Later on, we’ll analyze thoroughly the ways through which IP telephony can be materialized.

At this point, it is important to refer to certain basic operating principles of the IP telephony. Initially, the voice is converted to data packets. Flexible compression algorithms of the digitized system are used for this conversion, as they greatly reduce the required capacity for their transmission. Additionally, some specialized protocols that improve the quality and the continuance of the conversation are used for the transmission of the voice packets.

Just like every personal computer, every device has an IP address. It can also have a number similar to that of conventional telephones. Therefore, every voice packet includes the IP address of the receiver in order to reach the call recipient. The route that the packets follow is determined according with the rules of Internet routing.

As we said before, the IP telephony is not necessarily used only between VoIP terminal users, but also between users of classic telephony (either mobile or fixed one), through specific gates which IP telephony services providers dispose.

On our first occasion, we have two VoIP users. In that case, communication can exclusively be materialized through the Internet at no cost. Apart from the Internet connection, the utilization of the same application and the proportional equipment is essential so that vocal communication can be materialized.

On our second occasion, we have a VoIP user and a telephone user. In that case, no equipment is needed in either side. However, an active subscription to a VoIP services provider is required. Moreover, calls cost way less than normal long-distance calls.

On our third occasion, we have two phone users. In that case, an active subscription to a VoIP services provider is required. The provider uses the Internet as core network to transmit long-distance or international calls. The public telephone network is only used for the transmission of the call in the last mile. For this reason, the call of the cost is reduced.

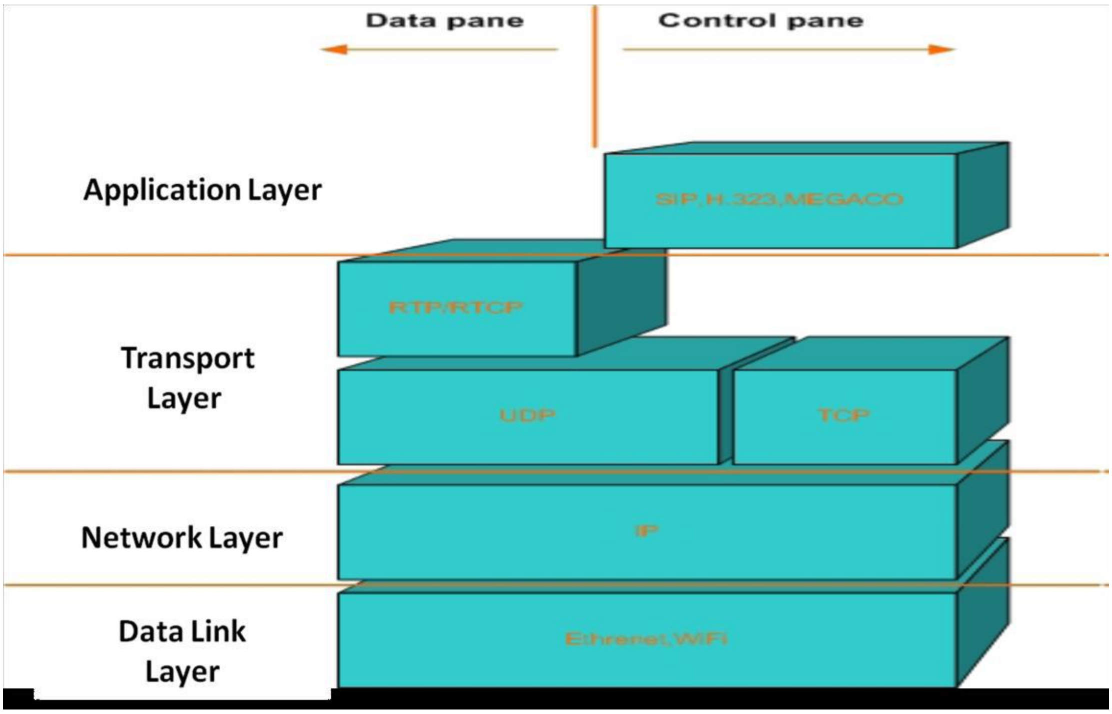
In this slide we can see an illustrative diagram of a unified corporate network of voice and data. We can easily distinguish the three ways to implement IP telephony.

One of the most crucial parts in an application is the usage of protocols and standards. IP telephony, as its name suggests, uses the Internet Protocol for voice transmission. This means that it uses an IP network as transmission medium, independently of the Data Link Layer and the Physical Layer. That means that the protocol that is used at the Network Layer is always the IP, whereas any compatible protocol, Ethernet or Wi-Fi can be used at lower levels. Depending on the implementation, there can be alterations at the Transport Layer and the Application layer. The function of a VoIP system in a session can be divided into two parts:

1. The creation and transmission voice packs and

2. The check of the VoIP call.

In the vast majority of applications, the Real Time Transport Protocol (RTP) is used rather than the User Datagram Protocol (UDP) for the transmission of voice packs at the Transport Layer.



For call control, session protocols TCP/IP are used, which are responsible for the beginning, the amendment and the ending of a VoIP call. The most widespread session protocols are Session Initiation Protocol (SIP), and the H.323 set of protocols. These protocols use the TCP and the UDP at the Transport Layer. Both protocols belong to the class of signaling protocols and do not constitute a complete VoIP application on their own.

SIP was the one that led the way for the rapid development of the VoIP. It was evolved by the Internet Engineering Task Force (IETF), and it embodies elements from two widely used protocols of the Internet, the Hyper Text Transfer Protocol (HTTP) which is used for net surfing, and the Simple Mail Transfer Protocol (SMTP), which is used for e-mail. As a matter of fact, the SIP borrowed the “client-server” architecture and the Uniform Resource Locators (URL) usage from the former, but it also borrowed the text format and the format of the headers from the latter. It represents a protocol of the TCP/IP application layer, and it is mainly used to install, modify and terminate bilateral or multilateral multimedia communication sessions, which consist of one or more terminals, like phone calls and video calls through the network.

The main four entities of the SIP protocol are the following: SIP User Agents (SIP clients), SIP Registrar Servers, SIP Proxy Servers, and SIP Redirect Servers. The entities above can receive and send SIP messages, for the installation of a VoIP call.

The Registrar Server accepts register requests and creates registries that correspond to the normal user addresses. This way the SIP Registrar Servers create databases with such registries for every active user.

The SIP Proxy Servers is the software that is implemented from the main hubs of a VoIP network and it accepts the requests that are submitted from the SIP User Agents for communication installation.

An SIP Redirect Server informs the users about proxy addresses or another user that will serve their request.

The SIP User Agents is the software that is implemented on the final users' devices, like smart phones, and the PCs during the installation of a VoIP call.

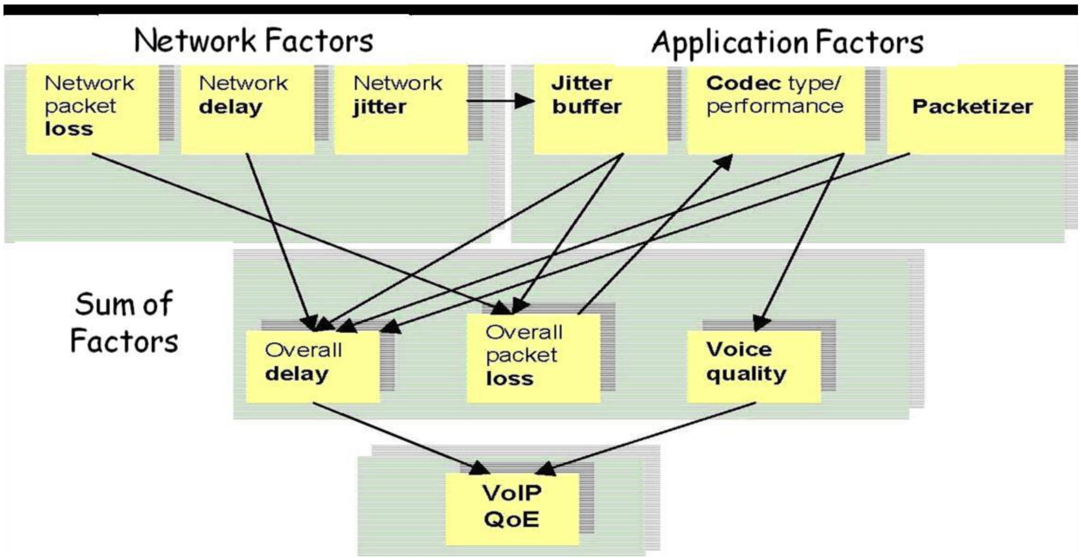
The tendency for transition from the conventional telephony to VoIP telephony is huge and continuously growing. Most VoIP system users have the demand that the quality of the VoIP service approaches the one of the traditional PSTN telephony that they were used to. This demand is a challenge by itself, as the VoIP, as an interactive multimedia real-time service that is transmitted above an IP network, can present many problems as far as its quality is concerned.

The main factors that affect the VoIP quality are:

1. The overall delay that greatly affects the quality of a VoIP conversation, mainly because of its interactive character, and

2. The quality of the voice signal that reaches the user.

The latter is affected by certain factors. To be more specific, there is:



- The fluctuation of the delay, Jitter, which particularly affects real-time applications, like the VoIP, as every voice pack must be presented to the user at a specific time moment.
- The Jitter buffer, which eliminates the fluctuation of the delay, but increases the overall delay and causes the rejection of certain packs.
- In addition, there is the Overall packet loss, which directly affects the assumed voice quality, since the initial voice signal is distorted with pack loss.
- The encoders, because when a signal is encoded and then decoded, it undergoes permanent distortion.
- Finally, there is the Packetizer, which groups a number of frames, forming the load of a packet, which is sent to its destination after the suitable headings in proportion with the protocols in use (e.g. RTP/UDP/IP) are added to it. The size of a packet is shaped proportionally with the number of frames it incorporates. As a result, the bigger the packet, the longer it takes to be transmitted, creating a larger delay in communication.

Other factors such as the noise on the voice signal, the echo, and the volume, can also affect the quality of the voice signal and the quality of the conversation, but to a lesser extent.

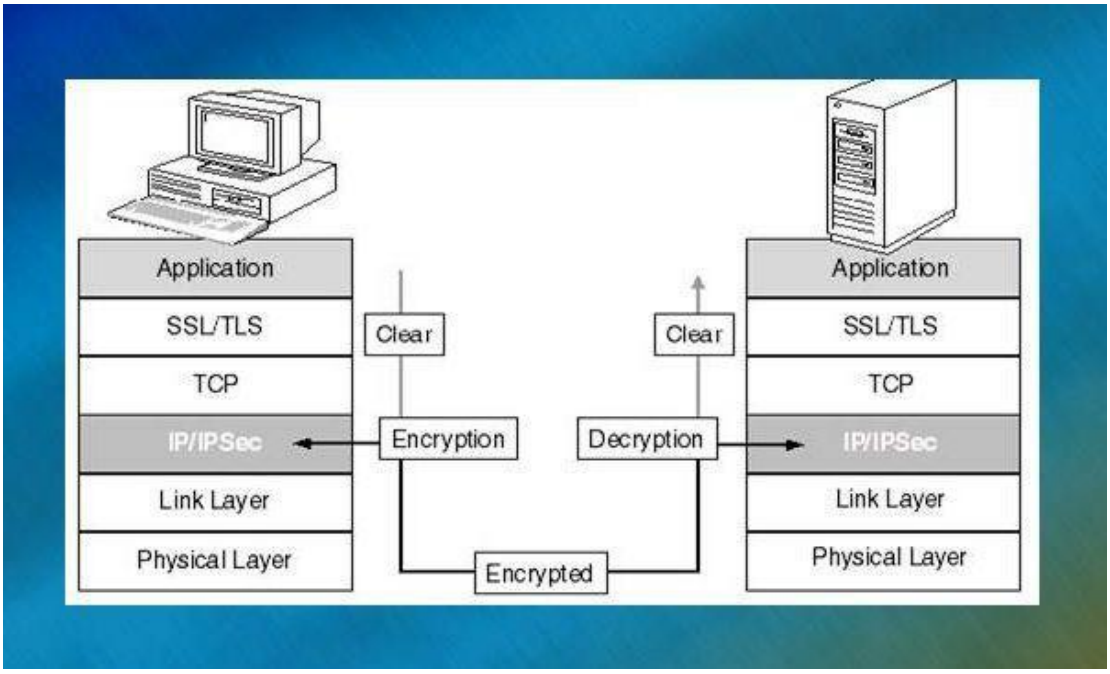
All in all, it would be a good thing to refer to the advantages of the IP telephony. The main advantage of broadband services is their cost. The VoIP providers charge local calls, long-distance calls, and international calls way less than the traditional telephony services. Some VoIP providers (e.g. Skype), grant communication in between personal computers for free using their software only. In addition, IP telephony provides conveniences that are usually not available with the traditional telephony service, such as the virtual phone number. In this way, users can select a phone number that has an area code different than the one he lives in. IP telephony is provided along with free services, such as conference calls or file exchange, which have an extra cost in traditional telephony enterprises. Even if there is already a normal telephony line, the VoIP is a good alternative that has low cost, and it can add an additional telephony line with extra characteristics. Last but not least, IP telephony can guarantee mobility and portability. Anyone can use VoIP services with broadband connection everywhere. The only thing you need is a computer and a handset, or simply a VoIP adapter phone.

On the other hand, there are some drawbacks. Some VoIP services do not function during blackouts. In addition, the quality is affected by the quality of the broadband connection and the computer performance. It is therefore essential that the internet connection is fast enough and the hardware as well as the software functions correctly and efficiently. As far as security is concerned, phone calls are easily accessible by third parties, because they diffuse into the Internet.

Up to this moment, we have only referred to the function of the IP telephony. Nevertheless, what is of actual interest to us is its implementation. In order to achieve this, everything should be supported by a Private Branch eXchange (PBX). PBX is a device which provides telephony services to a company, in the same way that a telephony services provider offers to the public. A PBX makes connections between the internal phones of a company, and in addition it connects them to the public telephone network via external lines. A PBX should provide high availability, reliability and efficiency with telephonic communication, while they should also offer advanced services, adding value to the telecommunication substructure of the company. The modern IP PBX, apart from their main function as phone call transport centers, they offer cheaper (in comparison to traditional PBX) services that have an additional value, such as Voice mail, Autonomous call distributor and Videoconference.

The main objective of this study is to examine the substantiation of a Private Branch eXchange with IP telephony in the armed forces. In order for that to be attained, its security should first be ensured. The protocol which can be implemented is the Internet Protocol security, also known as IPSec. The IPSec is a protocol of open specifications for the securing of the secrecy of telecommunications. It is based on the specifications that the Internet Engineering Task Force (IETF) has developed. The IPSec ensures the privacy, the wholeness, and the authenticity of communicative data in an IP network. It provides the essential equipment for the development of flexible security solutions within a network. In addition, it provides certification of authenticity and cipher at the Network Layer, and that's why it constitutes a vital part of the overall security.

The specifications of the IPSec define two new types of data to the packets; the certification heading for the data wholeness service supply, as well as the security abuse load, which provides identity certification and data wholeness. The communication parameters between two devices are also defined, which are the key management and the security correlations. The IPSec implements cipher and certification at the Network layer as we can see in the slide, offering in this way a security solution within the architecture of the network.



Thus the systems and the applications that are on the sides do not need alterations or adjustments in order to have the advantage of high security.

Since the ciphered packets look like normal IP packets, they can easily be routed via any IP network, like the Internet, without any alteration to the intermediate network equipment. The only devices that are familiar with cipher are the ones at the far-out parts. This characteristic reduces effectively both the implementation cost and the administration cost. The main advantage of the IPSec is its transparence with the applications. It functions at the Network Layer, thus it has no effect at higher levels to all intents. In addition, it is suitable for real-time motion guarantee, as the IP telephony requires.

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