Voice over IP: concepts, challenges, reality

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Introduction

Voice over IP (VoIP) stands for the transmission of voice signals placed in packets sent over data networks. Several terms are commonly used to describe this process: VoIP, Internet telephony, IP telephony and packet-voice. Unlike traditional telephony systems where a circuit-switched line with constant bandwidth is established between two parties of a phone call, in VoIP, the voice data is "packetized" in small units of user traffic. These IP[1]-packets are sent over a data network, e.g. the public Internet or a private network, to the destination where the voice signals are reconstructed.

Over the last few years, VoIP technology has emerged from laboratory experiments to widely used real-world applications. Although being a highly theoretical and complex field in the technical domain, VoIP has significantly emerged in practice. New applications and products have been invented and the quality and stability increase steadily. VoIP is becoming an important alternative to the daily used telephone system, not only in terms of economic aspects but also in quality of service for both, professional and private use.

In this project, we would like to present the key concepts and most important notions of VoIP. Various standards and recommendations have been defined and are applied in practice. In a first part, we provide an overview and brief description of these protocols in order to familiarize the reader with the basic vocabulary, terms and concepts related to this context. We will also cover the main challenges of VoIP from a technical point of view.

In a second approach, we consider VoIP technology from a social and economical perspective. We describe cost reducing solutions in corporate environments and provide insight into current and future fields of application for professional, educational and home use. A case study of a Swiss company gives a notion of the effective economical aspects of a corporation-wide VoIP deployment.

In spite of the fact that VoIP is relatively recent, it has become a serious branch of today's telecommunication market: various competitors offer devices and software for end users and corporate environments as well as for service providers. Already, a substantial number of service providers challenge the traditional telephony system offering reliable communication at low cost for national and international phone calls. We try to give a market analysis highlighting and comparing products and services and their operational fields described in sample setups.

Finally, we conduct a survey at EPFL and give an impression on acquaintance, interest, and usage of VoIP in the EPFL community. The result of this survey gives us the possibility to determine and understand the needs and expectations of a VoIP service but also reasons that prevent people from using it.

February 2005, Marc Stoecklin and Patrik Bless

[1]Internet Protocol. IP defines how information travels between systems across the Internet.

Chapter 1

Technical aspects, standards, challenges

This chapter aims at giving information concerning the use of VoIP. Furthermore, important standards and relevant associations are introduced. Designated associations and task forces regulate the evolution of VoIP and guide it in the sense of accepting and publishing new standards. Software and hardware manufacturers can therefore stick tightly to these publicly accepted terms and implement their commercial solutions. The development and introduction of a new technology is always linked to technical challenges. We would like to present some of them towards the end of this chapter.

1.1 Why Voice over IP

The old-fashioned circuit switched telephone network we tend to use every day has a couple of disadvantages. An important factor is the lack of integration of voice and data.

- New applications will demand a close link between data and voice. As a first example, video conferencing might be mentioned. Video conferencing allows to establish a voice connection while being able to see the person one is talking to. This type of software is becoming increasingly more important for multi-national companies. These companies require a secure and reliable channel without the possibility to intercept traffic or inject unwanted information. A close connection between data and voice is important and required also for document exchange (e.g. contracts in a board meeting) during a video conference.

- The use of VoIP can significantly simplify network structure and architecture even for smaller associations. Less equipment and cabling is used when voice is transported over the IP network. In addition, improved software products are able to facilitate the way the IP telephone is used (e.g. no need to transfer the telephone book to the phone, online administration of voice mail box, simplified telephone conferencing possibilities).

- VoIP does not demand network resources when no connection is active. This significantly reduces network load. In an ordinary phone network, network bandwidth is allocated to a user even when he is not talking. The idea is thus to provide bandwidth consolidation. Connections demanding more bandwidth can therefore claim other unused resources improving thus the overall performance. An additional factor consists of analyzing the speech and voice patterns. By removing redundant information and applying intelligent compression and coding algorithms, bandwidth requirements can again be reduced.

- An existing VoIP installation is easily scalable. Without having to register new phone numbers with a telecom operator, new IP addresses (especially with IPv6) can be used to attach a new phone to an existing network.

The factors mentioned beforehand aim at balancing the available bandwidth and reducing the cost for voice communication. Especially international calls are cheaper when using the internet backbone provided by internet service providers (ISPs) instead of the ordinary telephone network. Moreover, telephony companies (TELCOs) charge monthly access fees in order for the user to be allowed to use their local access facilities. These can eventually be decreased by reducing the number of telephone lines. Of course, inter-company communication has to be considered, as well. The enormous amount of internal phone calls can be established over the IP network avoiding thus the expensive installation of Private Branch Exchange[1] (PBX).

- To look up phone numbers and other information, IP enabled phones could be equipped with directory lookup services. These services are located on the web and the phone is able to connect to them without generating any additional cost.
- As another technical advantage the possibility of alternative routing has to be mentioned. As IP allows several ways to reach a target, the communication link becomes more reliable and thus more resistant against failures of a single TELCO.
- Finally, the limitations imposed by the Plain Old Telephone System (POTS) reduce the available bandwidth for speech to 64kbps. This is clearly not very much. Since VoIP transmits the whole spectrum, applications and devices are able to transmit high-fidelity voice.

There is still a great number of advantages that could be listed here. For different applications and uses, the one or other might be of particular interest. We limited ourselves to a number of commonly accepted improvements achieved by the installation of VoIP.

1.2 VoIP Protocols

Communication exchange always has to be guided and ruled by the means of protocols. These protocols describe the way two processes communicate with each other exactly. In our case, these two processes are VoIP end devices or software applications running on a personal computer. To gather an overview we introduce the two mainly used protocols for VoIP.

1.2.1 H.323

H.323 is a broad and flexible recommendation proposed by the ITU-T[2]. As a minimum, H.323 specifies protocols for real-time point-to-point audio communication between two terminals on a packet-based network that does not provide a guaranteed quality of service. The scope of H.323, however, is much broader and encompasses inter-network multipoint conferencing among terminals that support not only audio but also video and data communication.

The scope of Recommendation H.323 can be summarized in the following broad categories[1]:

- Point-to-point and multipoint conferencing support: H.323 conferences may be set up between two or more clients without any specialized multipoint control software or hardware. However, when a multipoint control unit (MCU) is used H.323 supports a flexible topology for multipoint conferences. A multipoint conference may be centralized where new participants can join all the others in the conference. This is the so-called hub-and-spoke topology. Or, a multipoint conference may be decentralized where new participants can elect to join one or more participants in the conference but not all. This approach will produce a flexible tree topology.
- Inter-network interoperability: H.323 clients are interoperable with switched-circuit network (SCN) conferencing clients such as those based on Recommendations H.320 (ISDN), H.321 (ATM), and H.324 (PSTN/Wireless).
- Heterogeneous client capabilities: A H.323 client must support audio communication; video and data support is optional. This heterogeneity and flexibility does not make the clients incompatible. During call set-up capabilities are exchanged and communication established based on the lowest common denominator.

[1]A subscriber-owned telecommunications exchange that usually includes access to public switched networks
[2]International Telecommunication Union -Telecommunication Standardization Sector (http://www.itu.int)

- Audio and video codecs: H.323 specifies a required audio and video codec. However, there is no restriction on the use of other codecs and two clients can agree on any codec which is supported by both of them.
- Management and accounting support: H.323 calls can be restricted on a network based on the number of calls already in progress, bandwidth limitations, or time restrictions. Using these policies the network manager can manage H.323 traffic. Further, H.323 also provides accounting facilities that can be used for billing purposes.
- Security: H.323 provides authentication, integrity, privacy, and non-repudiation support.
- Supplementary services: H.323 recognizes the huge potential for applications based on IP telephony and multimedia. It provides a basic framework for development of such services. As an example, two services – call transfer and call forwarding – have been specified in version 2.0 of H.323.

The current version of H.323 is H.323v5 which dates from July 2003. A list of new features introduced in version 5 can be found at Packetizer.com[3]. In the open-source world, people also started to implement the H.323 standard[4]. OpenH323 is published under the Mozilla Public License[5].

1.2.2 SIP

The following is an extract of Rakesh Arora's paper[2]:
The Session Initiation Protocol (SIP) is the IETFs standard for establishing VOIP connections. It is an application layer control protocol for creating, modifying and terminating sessions with one or more participants. The architecture of SIP is similar to that of HTTP (client-server protocol). Requests are generated by the client and sent to the server. The server processes the requests and then sends a response to the client. A request and the responses for that request make a transaction. SIP has INVITE and ACK messages which define the process of opening a reliable channel over which call control messages may be passed. SIP makes minimal assumptions about the underlying transport protocol. This protocol itself provides reliability and does not depend on TCP for reliability. SIP depends on the Session Description Protocol[6] (SDP) for carrying out the negotiation for codec identification. SIP supports session descriptions that allow participants to agree on a set of compatible media types. It also supports user mobility by proxying and redirecting requests to the user's current location. The services that SIP provides include [RFC2543]:

User Location determination of the end system to be used for communication

Setup ringing and establishing call parameters at both called and calling party

Availability determination of the willingness of the called party to engage in communications

Capabilities determination of the media and media parameters to be used

Handling the transfer and termination of calls

Components of SIP

The SIP System consists of two components [3]:

1. User Agents:
   A user agent is an end system acting on behalf of a user. There are two parts to it: a client and a server. The client portion is called the User Agent Client (UAC) while the server portion is called User Agent Server (UAS). The UAC is used to initiate a SIP request while the UAS is used to receive requests and return responses on behalf of the user.

[3] http://www.packetizer.com/voip/h323/whatsnew_v5.html
[4] http://www.openh323.org/
[5] The full license text is available at http://www.mozilla.org/MPL/MPL-1.0.html
[6] See section 5.5 of the mentioned document for a description of SDP

2. Network Servers:
   There are 3 types of servers within a network. A registration server receives updates concerning the current locations of users. A proxy server on receiving requests, forwards them to the next-hop server, which has more information about the location of the called party. A redirect server on receiving requests, determines the next-hop server and returns the address of the next-hop server to the client instead of forwarding the request.

1.3 SIP or H.323?

SIP and H.323 are the two predominant protocols used for inter-domain VoIP connections. There are a few other standards in the field, such as Cisco's Skinny protocol (SCCP), the IAX protocol of the opensource telephone system Asterisk or the Media Gateway Control Protocol (MGCP). However, these protocols are only used in LAN telephony systems and not over the Internet.

SIP is more lightweight in terms of coding since all information is coded using the omnipresent ASCII[7] standard. This renders SIP suitable for any kind of device without concerns regarding the support of different character sets. Furthermore, the protocol is fairly easy: A standard phone call requires only four header lines (To, From, Call-ID and Cseq) and three messages (INVITE, ACK, BYE). On the other hand, an implementation of H.323 has to support a number of supplementary protocols. Moreover, commands are transmitted in binary mode.

To sum up, both protocols have their drawbacks. While SIP is easier to use and less complex than H.323 also because of its similarity to HTTP[8], some protocol overhead might arise when implementing extension in other programming languages such as JAVA or PERL.

For the industry it is not yet very clear which protocol is going to be predominantly used in the future. Some providers even use both protocols to provide the freedom of choice for the customer. Nowadays, a slight preference for SIP is encountered when comparing some different providers in the EU.

1.4 Technical challenges

The benefits of voice transmitted in IP packets clearly outweigh its disadvantages. However, it should be pointed out that these disadvantages have contributed to the slow integration of the technology in both the home and the business domain. Among these disadvantages are:

Quality of Service (QoS)

Current networks have difficulty providing Quality of Service (QoS) and enough reliability of packet delivery.

Purchase costs

To decrease bandwidth consumption and to enable toll quality, voice compression algorithms and echo cancelation requires additional processing power that makes digital phones more expensive than analog phones.

Complexity

As a practical matter, VoIP cannot be deployed instantly everywhere. VoIP networks are more complex with respect to existing phone networks.

Network technologies

Current network technologies using NAT[9] and internal IP addresses make it difficult to locate VoIP phones and establish calls. Firewalls and zone management can block traffic to pass through a network.

[7] ASCII stands for American Standard Code for Information Interchange and is a way of coding 128 characters using 7 bits prepended with one parity check bit

[8] HTTP denotes the Hypertext Transfer Protocol which is used to fetch webpages and associated information (images, etc.) over the Internet

[9] NAT stands for Network Address Translation and is used to let different clients connect to the Internet using only one shared IP address

One of the most important factors for successful deployment of VoIP is the overall quality. To achieve a toll quality in phone conversations, reliable transport of the voice packets at a certain bandwidth is required. The Quality of Service (QoS) in VoIP depends mainly on

Low delay (latency)

Long delays (more than 400 milliseconds) make conversations difficult.

Accurate delivery

The transmission path must be predictable and quasi-static such that there is no large variation in the end-to-end arrival time of the packets.

Low packet loss

A voice packet contains 10 milliseconds of data, hence the loss of a packet is severe a problem when reconstructing the voice since it cannot be retransmitted later.

Bandwidth

To ensure that real-time data packets such as voice arrive reliably, a certain bandwidth needs to be available along the sending path.

Error minimization

Minimizing the errors in the speech coding algorithm.

Chapter 2

Social and economic aspects

As an effect of the sudden collapse of the New Economy and the DotCom bubble at the end of the 20th century, the telecommunication industry suffered difficult years. Nevertheless, there are clear signs that the VoIP technology is going be publicly accepted on the broad market and helps the industry find back new strengths.

The United States and Canada representing the largest market of deployment are closely followed by Europe and Asia[4]. However, there is a strong pressure towards low prices and low-tech end devices. Both, the corporate environment as well as in the SOHO[1] domain show an interest in user-friendly telephony equipment that are simple to manage and operate at low running costs.

In this chapter we present common fields of application in corporate and private use where VoIP is bound to replace conventional telephony nowadays as well as in the nearer future. Challenges with regard to the use by end users are discussed in section 2.2. We conclude this chapter by conducting a case study of a Swiss company.

2.1 Field of application

In most parts of the world, cost reduction for long distance phone calls (e.g. international) and the ability to maintain lightweight internal telephone structures are crucial arguments in corporate environments to migrate towards Voice over IP systems. However, the technology only becomes useful when groundbreaking applications meet the needs of customers. In this section, we try to highlight three of such applications. The deployment of PBX[2] based on Voice over IP in a corporate environment is basically interesting from an economical point of view. On the other hand, cable operators and service providers offer services to home users. New enterprises enter the telecom market and launch competition on the so called "last mile". Video telephony, multimedia enabled lectures and video conferences are a third field of application we will discuss.

These applications are the driving factors which allow manufactures to produce equipment, the operators to offer their services and the consumers to use them in order to increase their productivity. Once Voice over IP has fully come into the market, a more elaborate business model enables the technology to address the next generation of applications which will make the market grow further.

2.1.1 Reducing cost in corporate environment

IP based telephony applications offer attractive business models for professional customers today. One of the most important applications are the replacement of the traditional PBX, new client side applications using computers and the reduction of maintenance expenses for wiring and organizational modifications.

Today, PBX systems allowing a flexible telephone system based on Voice over IP technology are established on the market. They provide superior return on investment compared to traditional

[1]Small Office/Home Office. So-called SOHO products are specifically designed to meet the needs of professionals who work at home or in small offices.

[2]Private Branch eXchange. A private telephone network used within an enterprise. French: autocommutateur privé, German: Nebenstellanlage, see section 3.1.2

PBX systems. Despite the fact that initial equipment purchase costs are comparable, the setup and installation of VoIP PBXs turns out to be much less expensive since they use the existing data infrastructure such as LAN networks rather than separate voice wiring. Administration is also less burdensome because LAN and server administrators of the corporation's IT department are able to manage the system without the need for dedicated telephony technicians.

In Switzerland, some large and middle-sized enterprises but also administrative authorities have successfully replaced their infrastructure and are now using IP telephony without any loss of comfort but at running costs reduced by up to 40%. Some examples of such corporations and authorities are Helvetia Partia, Glarner Kantonalbank and Biel/Bienne municipal authority. Furthermore, in section 2.3 we present a case study of a VoIP migration at a Swiss steel trading company.

In the following, we discuss three possible schemes of application of IP-telephony systems within an enterprise. In the process of their integration, IP telephones are installed at every workplace and connected to the LAN. An employee will not observe any differences except the exchange of the telephone device. However, at the background, a call management system will be installed which is connected to the LAN, the PSTN[3] and possibly to the Internet in order to connect to other sites of an enterprise. A typical H.323 call manager implies a Multipoint Controller (MCU) and a Gatekeeper. A SIP-based setup requires a Proxy, Redirect and a Registrar Server (c.f. section 1.2).

Independent telephone systems

The simplest scenario is an enterprise which has only one site or uses a Voice over IP installation for each site locally. In such a case full communication structures are installed for each site individually as depicted in figure 2.1. Every site has its own call management system (e.g. Cisco CallManager) and a PBX that interfaces with the PSTN and handles external phone calls.

The PBX guarantees that internal calls, i.e. calls which have two participants at the same site, do not leave the site and will not be charged by the carrier. Therefore pure IP-telephony remains in-house. All other calls are routed via the PSTN; outgoing calls to other sites of the company, too.

Decentralized call management systems

Figure 2.2 shows a decentralized approach of call management: every site runs its own call management server. However, a so called Gatekeeper acting as a higher instance is required which performs access control and the resolution and management of phone numbers. At the same time, the Gatekeeper carries out the interconnection between all company networks over the Internet.

Interconnecting all sites with VoIP is very interesting for international companies. Internal phone calls are not charged by the telecom operator. Additionally, international phone calls might even first be routed to a site in the particular country via VoIP and then be passed to the local carrier to bypass high international tolls. However, this setup is considered rather expensive in terms of the hardware that has to be installed at all sites.

Several sites with central call management

The third scenario uses a call manager running at one site (c.f. figure 2.3). Normally, the call manager is set up at the headquarters or at the main IT department for management reasons and acts as call manager for IP phones of all sites. This structure allows central call management, however the total number of participants within the cluster is limited by the software or hardware. Cisco[5] foresees 2500 individual participants located at any number of sites for its call manager.

The main advantage of this approach is the saving of expensive call manager hardware and their maintenance at all branches. The design of the scenario should be supported by dedicated back-up lines to avoid overloads.

[3]Public Switched Telephone Network. The conventional telephone system.

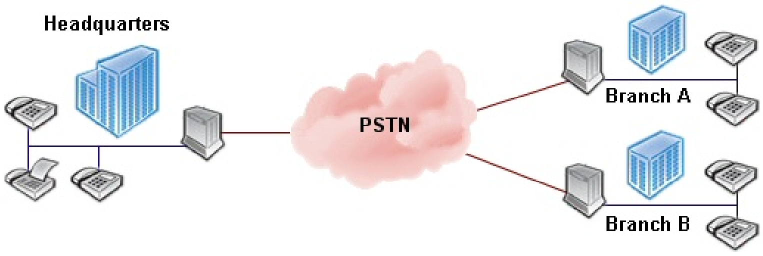


Figure 2.1: Independent call management systems for each site

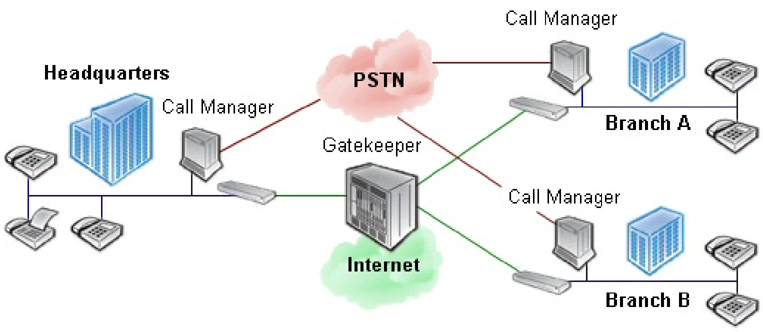


Figure 2.2: Decentralized call management systems interconnected via a single gatekeeper

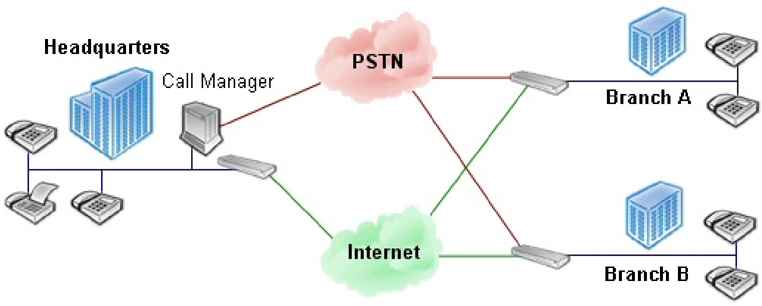


Figure 2.3: Centralized call management system

2.1.2 Home user benefits

A home user might experience nearly the same benefits already mentioned for the corporate environment. VoIP also guarantees low rates for home users. A user might thus stay in contact with relatives living far away and even exchange data (e.g. images) while they are talking. With VoIP getting more and more powerful, even connections to ordinary fixed line phones are possible at very low rates. As an example, we can consider Skype[4] that allows to do phone calls to Japan at rates that are even lower than the ones charged for an ordinary connection in Switzerland. This is really amazing and the home user might also be persuaded by these benefits.

An additional advantage consists of inter-building communication. Connections from room to room are possible over a standard Ethernet link and no additional wiring is needed.

A solution widely used for commercial home user VoIP is the PacketCable™ telephony architecture which runs over the Cable TV systems. To use the service, a user subscribes for it at his TV cable provider and installs a special modem at its TV connection. Traditional telephones may be plugged in to this device and can be run as usual. Most of the providers allow porting of the previous fixed line telephone number to the new subscription, thus no announcements to friends or relatives are necessary. Current actors and providers in this field are described in section 3.2.

Another possible solution for a home user is the subscription at a SIP provider offering to make phone calls at low rates over the Internet via the provider's remote servers. The home user either uses an IP telephone with SIP capabilities or an adapter translating traditional phone signals (SS7) to SIP packets similar. In sections 3.1.1 and 3.2, we take a closer look at available hardware products and SIP providers respectively.

2.1.3 Video Telephony and Conferencing

A third important field of application is video telephony and conferencing over VoIP. Video telephony includes the use of packetized voice and images and is said to be the next generation of communication for both the home and business domain. Coupling the recently deployed UMTS video telephony with Voice over IP video service allows mobile and fixed line visual communication.

A second interesting use of transmission of video signals in VoIP are interactive lectures and conferences. In the following, we present one of possible envisioned scenarios for a lecture[6]: A professor might give a lecture in the United States while students from all over the world may join the lecture and interactively participate. Running a special application on a PC or a projection at a university, the current slides might be shown as well as the professor's talk. Simultaneously, the professor is able to provide handouts and exercises over the connection or to perform a quick poll among the students. Interaction from students, e.g. questions or answers, can be handled in many ways. Normally, the conversation in a lecture connection is one-way, i.e. the professor talks to the students, however, the professor may have the opportunity to give the microphone to a student (which is "raising his hand" interactively) to allow a question/answer. Depending on the professor's choice, the voice of the student is either heard by the whole class or by the professor only ("Spare"). The students may also have the possibility to use chats and discussion boards the discuss problems during and after the lecture. In this sense, remote exercise sessions and team projects may also be possible.

2.2 Implementation challenges

The acceptance of a VoIP system for home users depends on different factors than for corporate users. Home users tend to like the systems they are used to work with (for example at their work place). Independence from a computer is also a fairly important issue since one might not want to sit near a computer all the time to do a phone call over the Internet. In addition, the ease of installation and maintenance of a VoIP solution plays an important role. VoIP should be easy to install and should not create any problem nor inconveniences for the user. This includes an easy, straight-forward setup and a simple way to call another person. The number of devices needed is also important. A home user might not like several oddly-looking devices in his/her kitchen,

[4] refer to section 3.3

living room or bedroom. This implies the independence from computers. VoIP products are only accepted on the broad market when a true independence from computers can be guaranteed.

Another issue are costs. Home users are very sensitive and tend to compare between different ways of achieving the same result. Therefore, clear and transparent costs have to be offered to the customer probably even by a company the user already trusts. The need to switch to a completely unknown company might make users hesitate.

2.3 Case study: A Swiss steel trading company

Due to company policies we treat the names and locations confidentially.
The given figures represent a typical example.

The presented company takes a leading position in the distribution of materials all over the world. The Swiss company that enjoys a long tradition has at its disposal not only numerous regional warehouses and selling points, but also a very broad assortment of goods.

During the preparation of this project we had the opportunity to have a look at an actual evaluation of VoIP technology in a corporate environment. The IT responsible of the allowed us to catch a glimpse at working documents and cost comparisons that showed the actual benefits of VoIP versus the ordinary telephony system.

The company's IT department received requests to renew old telephony systems from three of its over two dozens affiliated societies. These societies being located in Site 1, Site 2 and Site 3 already dispose of a standard network installation for office use (except Site 1). This facilitates and makes an installation of VoIP cheaper since no additional wiring is required. However, in order to support a VoIP installation in all affiliated societies, some additional hardware and software has to be purchased. The chosen VoIP provider Swisscom Solutions proposed a setup that requires an investment of another CHF 159'000.- in order to purchase additional hardware in the IT center in Call Manager, cf. section2.1.1 "Several sites with central call management"), cover the necessary expenses for engineering work and acquisition of hardware in the affiliated societies. The total amount of CHF 360'000.- is therefore astonishingly low considering the fact that the basis for further VoIP installations in the remaining affiliated societies is now set up. New societies can be intergrated in the future without any major changes of the overall structure of the network. As listed in Table 2.1, the cost of telephony equipment in a VoIP environment is considerably lower than the cost of an ordinary telephony system.

	conventional	**VoIP**
Site 1		
Equipment	CHF 79'000.-	CHF 42'000.-
Infrastructure	CHF 57'000.-	CHF 55'000.-
Total	CHF 136'000.-	CHF 97'000.-
Site 2		
Equipment	CHF 55'000.-	CHF 56'000.-
LAN-S Hardware		CHF 38'000.-
Total	CHF 55'000.-	CHF 94'000.-
Site 3		
Equipment	CHF 70'000.-	CHF 48'000.-
LAN-S Hardware		CHF 22'000.-
Total	CHF 70'000.-	CHF 70'000.-
IT center		
Call Manager (redundant)		CHF 39'000.-
Engineering/ concept		CHF 60'000.-
Total	CHF 70'000.-	CHF 99'000.-
Grand total	CHF 261'000.-	CHF 360'000.-

Table 2.1: Cost comparison

In addition, there are other particular advantages that made the board agree on this solution. Some of them are listed as follows:

high scalability IP telephones are more flexible and scalable the traditional PBX equipment

integration of applications IP telephony systems are easier to expand due to the use of standard protocols (CTI[5], Unified Messaging, XML, etc.)

high availability high availability can be achieved with redundancies (distributed network components, rescue systems)

reduction of operation cost work places can be moved easily due to central management, shared usage of human and network resources

Some more technical reasons contributed to this decision as well:

backup integrated in existing concept: 200–500 MB storage capacity required

anti-virus only one licence required

Windows 2000 patches work load of only one hour per month

CTI dialling initiated by messaging software (e.g. Microsoft Outlook)

WebDialer dialling initiated by URL (intranet phonebook)

integration in CRM[6] same technology used (Server, IP, etc.)

Since not all affiliated societies are changed to the new technology, the old systems continue to exist and special attention has to be given to the future change of old structures to VoIP in the remaining societies. VoIP technology is able to handle this challenge perfectly since the system is easily scalable to a larger environment. The main points in favor of VoIP concerning scalability are:

- initial phase with only one Call Manager is possible
- scaling effect when migrating other sites – investment limited to office phones, Call Manager is used centrally
- easy migration of remaining sites – VoIP used in parallel to existing system; seamless abandonment of old system
- central management – cost, efficiency, standardization
- inclusion in existing Trouble Ticket System – Fault Management Data

Obviously, the initial cost of a VoIP installation is higher compared to the ordinary telephony system. However, future expansion and benefits clearly support the change to VoIP. Due to the mentioned facts, the company decided to change their infrastructure to VoIP. The changes are still ongoing and the process of integrating all the societies has not finished yet.

[5]Computer Telephony Integration

Chapter 3

VoIP products/providers and the market

In this chapter we would like to give a brief survey over products and providers acting on today's market in the Voice over IP domain. The following descriptions do not claim to be exhaustive but they are intended to give a idea of the situation and available products at the moment. In the first part we describe hard and soft telephones and switching devices designed to be used with Voice over IP, both with an end-user and a service provider aspect in mind. In section 3.2 we focus on the current Voice over IP offers and services by providers and carriers.

3.1 Available devices and software

As Voice over IP is a relatively recent technology, a lot of considerable development in software and hardware has been made during the last few years. In order to analyze current market products, they have to be categorized into two different domains, the user domain and the processing and service domain. The user domain includes hardware IP phones and adapters as well as PC based software phones. Gateways, gatekeepers and private branch exchange (PBX), both software and hardware products used by service providers, are located in the processing and services domain.

3.1.1 End-user products

In the end-user domain, there exist mainly two possibilities to connect to a VoIP provider: the end-user either possesses a SIP-based phone or he might plug in his traditional analog phone in an adapter which then performs the conversion of the conventional signalling to VoIP. As a third alternative, software telephones installed on a computer enable the end-user to utilize additional features such as video telephony, file transfer or even to assist to lectures and conferences all via VoIP.

The "hard" telephone devices

Of course, the most relevant product for the Voice over IP end user is the telephone device he needs to make his daily calls. Currently there exist quite a few mature telephones with SIP capabilities that look similar to traditional telephones. It is obvious that the VoIP telephone devices are intended to be as simple as traditional phones in their handling. The telephone devices communicate directly with a Voice over IP server, gateway or another phone, thus they do not require any personal computer running in order to make or receive phone calls. They can be used independently – all that is necessary is an Internet connection. While PC based software solutions (see section 3.1.1) are cheaper (or even free), a hard phone is the best solution for IP telephony.

In general, so called "hard phones" are equipped with an Ethernet (RJ45) port and directly connected to the Internet via a router or a network switch. "Cordless Hard phones" have as well an IP interface at their base station but have independent hand sets. Self contained phones with a built-in modem instead of the Ethernet port are another option – they will connect through

Figure 3.1: "Hard" phone devices: conventional wire phone (Swissvoice IP10S), WiFi phone (VTech ip8100-2), video phone (Vonage F-1000)

the modem with a dial-up Internet service to a remote VoIP server. These phones are popular in countries where there is little broadband infrastructure available yet.

However, evolving technologies enable even more comfortable solutions: self contained phones with a built-in WiFi[1] tranceiver[2] go into the "WLAN or WiFi Phones" section. Together with a wireless access point, these phones allow the same freedom as cordless phones used up to now, except that they can also be used at public HotSpot locations, e.g. in hotels, restaurants, stations or universities.

Some of the most important telephone devices on the market categorized by type are presented in Table 3.1.

However, when analyzing the protocol specification and data sheets of products, it becomes obvious that in a VoIP phone sophisticated hardware and software is required. The phones need to have network interfaces (RJ45) instead of analog connectors, they have to cope with various network protocols and characteristics such as TCP, UDP, DHCP, NAT and PPPoE to be able to be put in place within a private network but also directly via a broadband connection like ADSL. At the same time, the signal processing part is more complex, e.g. devices must be equipped with jitter buffers, echo cancelation algorithms and voice activity detection discussed in section 1.4. Finally, several expensive cryptographic mechanisms are required during the SIP authentication steps.

Digital/Analog adapters

A second solution for the end-user is to install an Analog/Digital Telephone Adapter (ATA/DTA). An ATA is connected to the LAN/Internet via an Ethernet interface. Simultaneously, it provides at least one telephone port (FXS[3]) used to connect a conventional telephone. The ATA sets up the communication and performs the conversion between the telephone and a remote VoIP server using VoIP protocols such as SIP, H.323, MGCP or IAX.

As analog adapters communicate directly with a VoIP server, such an installation does not require any software to be run on a personal computer. Even an Internet subscription is not mandatory as there exists ATAs which have built-in dial-up modems connecting directly to a remote server via the existing phone line (FXS to Dial-up adapters). This is especially interesting for using VoIP to make international phone calls at local phone call charges.

A typical example for a simple ATA is Cisco Systems ATA-18x series (e.g. ATA 186 or ATA 188) which provide two regular analog telephone connectors and a 10/100BASE-T Ethernet interface to connect to the IP network. The adapters support H.323, SIP, MGCP and Cisco's proprietary SCCP[4] CallManager protocol.

AVM Fritz!BoxFon is a more sophisticated ATA device. It provides a combination of ADSL modem/router and Voice over IP PBX system in one box, i.e. it provides broadband Internet access

[1]WiFi stands for Wireless Fidelity, Wireless LAN (IEEE 802.11)

[2]transmitter and receiver

[3]Foreign Exchange Station. The FXS interface (RJ-11 connector) provides standard phone signaling for POTS, e.g. ring, voltage, dial tone etc.

[4]Skinny Client Control Protocol

Hard phones: Private		
3Com 3101 Basic	SIP, NBX (native 3COM)	
Giptel IP phones (G100)	SIP	$ 89.90
Grandstream BT101	SIP	$ 94.90
Siemens optiPoint 400	SIP, H.323	$ 440
Swissvoice IP10S	SIP, H.323, MGCP	$ 194
Hard phones: Business		
3Com 3102 Business	SIP, NBX (native 3COM)	
Cisco Systems 7940 series	SIP, Cisco CallManager, SCCP	$ 289.90
Snom 190	SIPS, GSM; Linux based	$ 259.90
Hard phones (Voice and Video)		
8x8 Paket8 VP	SIP based video phone	
D-Link DVC-1000	H.323 based video phone	
Wooksung WVP-2000	SIP, H.323; touch screen	
Cordless hard phones		
VTech ip8100-2	cordless broadband phone, 5.8GHz	
Dial-up hard phones (built-in modem)		
Dlink DPH-70	SIP or H.323; built-in 56k modem	
DialAnywhere Phone	SIP, H.323, MGCP; built-in 56k modem	
WLAN or WiFi phones		
Cisco Wireless IP Phone 7920	SCCP only, but SIP announced	$ 500
Hitachi WIP-5000	SIP	$ 250
Motorola CN620	supports GSM and WiFi	
Vonage F-1000	SIP	
Zyxel Prestige 2000W	SIP handset	$ 250
Soft(ware) phones		
DIAX	IAX protocol	
GnomeMeeting	H.323; open source, VoIP/video, Linux	
Nikotel	SIP; Windows, Mac	
SIP Communicator	SIP; Java based	
SIPhone	SIP, H.323; Windows, Linux, Mac, PocketPC	
Shtoom	SIP; Windows, Mac, Linux	
VLI GPhone	SIP; Windows, PocketPC, Palm	
Xten X-Lite/X-Pro	SIP; Windows, Mac, PocketPC	free/$ 50
Soft phones (Voice and Video)		
Apple iChat AV	proprietary, non-standard; Video, Audio, IM	
Windows Messanger	supports SIP; IM, Video, Audio	
Nortel SIP Multimedia	SIP PC client	
SIP Communicator	SIP; Java based	
TABLETmedia iFON	voice and video for mobile PDAs	
Siemens Suisse SA	SCS client for SIP	

Table 3.1: Telephone devices on the market.

Figure 3.2: Analog Telephone Adaptors: Cisco ATA 186 (left) and AVM Fritz!BoxFon (right)

via ADSL and telephony over Internet or the conventional telephone line at the same time. Since the Fritz!BoxFon is connected to the telephone line for ADSL access and has a second telephone (ISDN or analog connector), it is able to route outgoing telephone calls either via the ISDN/analog phone line or for economical reasons (e.g. international calls) over the Internet via the ADSL line.

PC based "soft" phones

A soft phone is an IP-telephone software. It can be installed on a personal computer or a mobile PDA and work the same way as a "hard" IP phone. Soft phones require appropriate audio hardware to be present on the personal computer they run. This can either be a sound card with speakers or earphones and a microphone, or, alternatively a USB phone set. Soft phones are inferior to hard phones but cheaper to obtain, many are available as a free download. Unlike hard phones, soft phones often come with video support which allow the user to communicate with more comfort.

Soft phones are basically used by persons who want to gain an impression of Voice over IP without spending a lot of money for a hard telephone. Often providers offer soft phones bundled in the subscription package such that the service can be used immediately. Another important field of application is the use of soft phones on mobile PDAs[5] such as PocketPCs or Palms. Nowadays, these devices are equipped with Wireless LAN transceivers and thus a soft phone running on a PDA is similar to a WiFi phone.

The most popular soft phone is Skype whose service we discuss in section 3.3. Other important IP telephone softwares are Microsoft's Windows Messenger, Apple's iChat AV or Xten's software suites. Xten's example shows that most vendors offer a free version (X-Lite) of their software which has limited capabilities. The full version X-Pro ($50) has about the same functionality as a small business phone. Xten offers also Video SIP soft phone application called eyeBeam ($60).

Figure 3.3: Soft phones: Skype running on Linux, Xten X-Pro on a Pocket PC, Apple iChat AV

[5]Personal Digital Assistant, a handheld computing device

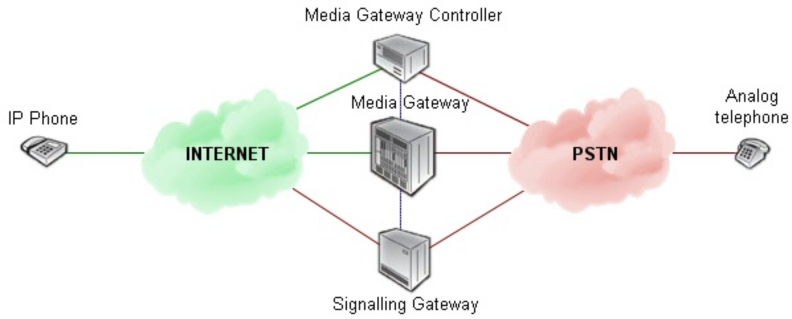


Figure 3.4: Bridging between the IP network and the PSTN

3.1.2 Processing and Service domain products

On the other end of a connection of Voice over IP telephones remote servers, gateways and PBXs are put in place. These devices are responsible for processing the incoming and outgoing phone calls, do the routing and also possibly conversion of signalling.

In this section we will consider PSTN Gateways which allow to forward Voice over IP calls on the conventional telephone network and PBXs which are typically employed to perform internal call management in companies.

PSTN Gateways

Voice over IP phone calls, propagating in a packet based manner on the Internet, need to be translated to the signalling used in conventional PSTN networks if a user wants to reach a traditional phone. A media gateway, also commonly referred to as VoIP gateway, is a device which performs bridging between the Public Switch Telephone Network (PSTN) and the VoIP telephone network. A basic setup of participating devices is depicted in Figure 3.4.

A *media gateway* (MG) provides translation of audio as a call passes across the boundary between two systems which use different encodings. It converts the digital voice data (codecs like ITU G.711, ITU G.723.1, GSM etc.) from the packetized IP network call to Time Division Mulitplexing (TDM) encoding used on conventional voice circuit. This means that its main task is the compression and decompression of voice signals. The *media gateway controller* (MGC), also referred to as call agent, is the central component in a transmission. It is responsible for handling and management of all resources. It takes care of the setup and teardown of calls, the control of communication connections and the media gateway via the MEGACO protocol. Additionally the MGC performs billing. The *signaling gateway* (SG) as the third essential component handles the termination and emulation of SS7[6] connections which are used in PSTN. On the other hand it translates SS7 events into a VoIP compatible format.

Several vendors like Cisco Systems Inc., 3Com or Nuera (among others) sell media gateways, signaling gateways and media gateway controllers. Nuera's ORCA BTX series provide robust telephony and IP interfaces, voice processing and network management at a low deployment cost for carrier-class telephony operations. The gateways support up to 68 T1/E1[7] interfaces (which corresponds to 2040 voice ports).

PBX (Private Branch eXchanges)

A PBX is a private telephone network within an enterprise. Users of the PBX system share a certain number or external lines leaving the private network towards the public switched telephone network (PSTN) for external calls. Most medium-sized and large companies use a PBX for economical reasons. Internal calls are kept within the company and are not routed to the telecom provider.

[6]Signalling System 7

[7]dedicated point to point, high capacity digital service capable of transmitting data at speeds up to 1.544 Mbps, offers 24 channels, each at 64 kbps

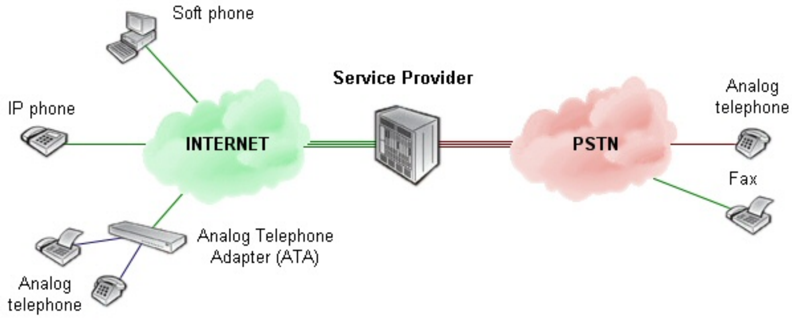


Figure 3.5: Interconnection between the VoIP customers, service provider and the PSTN.

Additionally, a PBX provides useful features such as call forwarding and transfer, short internal numbers (e.g. suffices of external numbers) or automatic call answering.

In general, a PBX is located at the enterprise which either runs itself the maintenance and setups (e.g. IT department) or hires specialists from professional suppliers (e.g. Swisscom Solutions in Switzerland). IP Centrex is another model where call platforms and PBX features are hosted at the service provider location. The business end-users connect via IP to the provider for all phone services.

Today, many PBXs based on Voice over IP technology are available. One kind of implementation is a dedicated trunk providing the necessary connectors both on the IP networking and the PSTN side. As a reference product for a medium-sized company, we mention 3Com's NBX 100 Communication System which supports up to 200 devices (lines and stations) including up to a maximum of 100 PSTN central office lines.

A second kind of PBX, which are widespread today, are PC-based software PBXs. Probably the most important actor in this field is Digium's Asterisk, an open source telephony switching platform. Asterisk runs on Linux and Unix platforms and provides interconnection between IP-based telephony VoIP standards like SIP, IAX[8], MGCP and H.323 and hardware that connects to the PSTN like Zaptel, ISDN BRI[9] and PRI[10] and other devices. Additionally it enables a set of functionalities like voicemail with directory, call conferencing, interactive voice response and call queuing.

3.2 Providers and actors

The basic field of activity of a Voice over IP provider, also called carrier, is to offer IP based call processing units similar to the fixed telephone line carrier. Moreover, carriers need to provide additional functionality and features such as acting as a gateway to the Public Switch Telephone Network (PSTN) to enable calls to traditional telephones, provide a location register and proxy service. As depicted in Figure 3.5, the carrier is located on the boundary between the Internet and the PSTN to interconnect both networks. Hence, it is able to forward calls originating from one of its customers either to another VoIP customer via Internet, to a gateway of another carrier (also via Internet in general) or to the traditional telephone network. On the other hand, he receives phone calls entering from the PSTN heading to one of its customers. These calls are translated appropriately and passed on to the the receiver.

In the last year, the number of providers offering Voice over IP services on the European market grew considerably. Telecommunication providers, TV network companies and new start-ups decided to take their chance in this evolving domain. Roughly they can be segmented into four categories which are described briefly in the following.

[8] Alternatively used protocol similar to SIP and H.323

[9] Basic Rate Interface. A type of ISDN service which comes with two B- Channels at 64 kbps each and a D-Channel at 16 kbps.

[10] Primary Rate Interface. A type of ISDN service that includes 23 B-Channels and a D-Channel.

Table 3.3 at the end of this section compares the most important players in the Voice over IP home user domain in Switzerland.

3.2.1 Internet/TV Cable providers

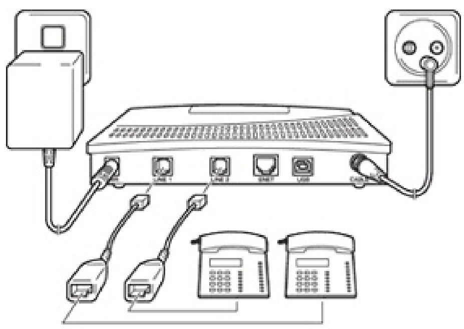
Some of today's Internet providers offer their customers packages containing Internet access combined with VoIP services or even VoIP services only. Normally the Internet connection is run either via the existing telephone line on ADSL or via the TV cable. Recent products allow the use of the electric power line or satellite links to offer fast and reliable Internet connections. The subscriber of such a combined package does not need a conventional fixed telephone line but uses the Internet for all incoming and outgoing phone calls. It is widely assumed that this solution points the way to the future: every customer has a single connection for all applications: telephony, Internet access and TV.

Cablecom, the largest cable communications provider in Switzerland, has launched "cablecom digital phone" on July 1, 2004, offering IP telephone services over their TV cables (see Table 3.2).

Other examples of leading Internet providers offering integrating VoIP services are Broadnet-Mediascape[11] and QSC[12].

Cablecom's "digital phone"

- low monthly line rental fees (subscription fee) of CHF 20.-
- Swiss fixed line calls during evening, night and weekend calls are completely free of charge
- low per-minute rates for calls during day: 3 cts/min (Swisscom 8 cts/min, Sunrise 6.9 cts/min)
- integrated voice mail which can be listened at any time even from road and the Internet
- no computer, Internet access or special phone required



A cable modem similar to an ATA presented in section 3.1.1 is connected to the TV cable network of Cablecom. The modem, providing connectors for a computer and phones, performs the conversion between packetized voice from the remote gateway and analog signaling for conventional telephones.

Table 3.2: The principle of Cablecom's "digital phone"

[11] http://www.broadnet-mediascape.com/
[12] http://www.qsc.de/

3.2.2 Internet and telephone providers/reseller

Some Internet Service Providers (ISP) offer VoIP products as a supplement to their existing offers. The fixed line phone access is not replaced by IP telephony but their areas of responsibility are separated for specific purposes. The client decides which (type of) phone calls are effected via fixed telephone line and VoIP. He may place value added calls rather to the fixnet line whereas for national and international calls he may use VoIP. This is normally done automatically by using a dedicated hardware device[13] which is connected to the Internet and telephone network at the same time. Incoming phone calls are received over the normal telephone network as before unless they are not from a VoIP user.

In Switzerland, the ISP and telecom provider green.ch offers Voice over IP solutions which are generally combined with a broadband Internet access, e.g. ADSL.

3.2.3 SIP providers

The third category of providers offer pure SIP based services. As these providers are not involved in the Internet service, the customer is free to choose his Internet Service Provider (ISP). Outgoing calls are directed to gateways in the Internet which route a call to the destination or, if it is a call to a conventional telephone line (fixnet or cell phone), acts as an interconnection to the public switch telephone network (PSTN). Incoming phone calls from the PSTN reaching the gateway are translated to SIP and forwarded to the destination over the Internet.

Current trends[14] show that SIP providers team up and interconnect their services to be more competitive and attractive as against conventional carriers. In most cases IP-to-IP phone calls are free of charge.

Typical providers are Sipphone[15], Sipgate[16], Nikotel[17] and e-fon.ch[18], a Swiss company which specialized on business and private Voice over IP services.

3.2.4 Providers using proprietary systems

Some providers do not use the standard based SIP VoIP technology but develop their proprietary standard. Despite the overall quality of most of those systems is in the order of the standard based one, the drawbacks are obvious. Interconnection to other systems are generally not offered and the customer is at the providers' mercy: the development and improvements fully depend on a single institution and not to a open standard body. However, being free from standardization, the systems may offer new features which are not possible at the current state of the VoIP standards or which are still in draft.

Some of the well known proprietary systems offering Voice over IP are Skype, Apple iChat AV[19] or SquidCam[20].

3.3 The bad guy: Skype

Skype is a new free computer software that has been published in August 2003. Since the launch of the beta version, the client has been downloaded more than 48 million times. The look-and-feel of the software is closely related to well-know instant messaging clients and thus very easy to get used to. After signing up for a new account where a username has to be chosen, the new participant can immediately start using the system. It is possible to add other users by searching for them in a directory or simply by adding their name to the buddy list if their username is already known. The users are then shown in a list where it is possible to get to know, whether a user is currently online or absent. Call establishment is very simple. A connection is created only by double clicking on the user.

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[13] normally the Internet modem/router with VoIP extensions, e.g. AVM Fritz!BoxFon
[14] http://www.sipgate.de/user_interface/newsletter/sipgate_PM_20041208.pdf
[15] http://www.sipphone.com/
[16] http://www.sipgate.de/
[17] http://www.nikotel.com/
[18] http://www.e-fon.ch/
[19] http://www.apple.com/ichat
[20] http://www.squidsoft.com/squidcam/

Provider	Cablecom	green.ch	e-fon.ch	Nikotel	Sipphone
Product	digital phone	SOHO analog	sip-phone	Nikotalk	MySipPhone
Protocol	proprietary	SIP	SIP	SIP	SIP
Bandwidth	irrelevant	15-20 kbps	30 kbps	22-80 kbps	80 kbps
Hardware	analog phone	analog phone	analog/SIP/ soft phone	analog/SIP/ soft phone	analog/SIP phone
Operating systems for soft phone	-	-	Windows, MAC OS, Linux, Pocket PC	Windows, Mac OS	Windows, MAC OS, Linux, Pocket PC
setup fee	-	CHF 139.-	CHF 39.-	-	-
monthly fee	CHF 20.-	CHF 20.-	CHF 14.-†	free	free
internal calls	-	free	free	free	free
fixed line	3 cts‡	4 cts	3 cts	4.5 cts	5.5 cts
cell phones	45 cts	49 cts	45 cts	35 cts	36 cts
international$^\diamond$	10 cts	9 cts	5-10 cts	4-12 cts	3-15 cts
international cell phones	10 cts	49 cts	43 cts	38-52 cts	33-62 cts

† including 100 min free charge on Swiss fixnet
‡ 07.00 - 19.00, free of charge on weekends and nights, except cell phones
$^\diamond$ France, Germany, Italy, UK, USA, Canada

Table 3.3: Provider comparison (Sources: [7], [8], [9], [10])

Skype is based on a stream of UDP packets that are directly aimed at the destination user. However, as mentioned in section 1.3, NAT plays an important role and a VoIP system has to be able to cope with this. Skype uses a transparent approach. The user does not have to change any settings when connecting through a NAT box. The configuration is automatically realized and the system takes this into account when establishing connections.

Connections between users that are not behind NAT boxes or protected by firewalls is straightforward since the RTP stream is directly aimed at the target user. In the other case where a user is behind a NAT box, the connection is relayed by a server. This way of communicating with each other allows both users to keep their ordinary Internet settings.

Reaching an ordinary phone

Skype allows to connect to ordinary phones around the globe at constantly low rates using a service called SkypeOut. The user can buy credits and is then allowed to establish connections to common fixnet telephones. Instead of double clicking a username, that user can thus enter the phone number (including the international prefix) he tries to reach. Generally. this works very well and presents a way of calling people that do not use Skype. However, sometimes rather annoying delays might be experienced when using SkypeOut. For international connections this might not be a major issue since quality has always been quite low but for intra-continental communication people might still prefer the old telephone system.

Why is it bad?

Skype is available at no cost and also connections between two Skype users are free of charge. Also because of the easy configuration and the availability for different platforms (Linux, Mac, Windows), Skype presents an important alternative to other VoIP services. However, Skype does not use the two protocols presented in section 1.2 but has its proprietary protocol. Therefore, interoperability with existing providers is not possible and other VoIP providers are not able to profit from the success of Skype. This makes Skype a bad guy in terms of business but a preferable application for the user.

Chapter 4

VoIP at EPFL - a survey

Doing a project concerning a new technology becomes very interesting when including the link to real users. A large number of improvements and positive factors do not render a technology accepted by the public. Our intention was to get to know the acceptance and usage of VoIP and its derived products at EPFL. To reach the users, we created a small survey to facilitate user interaction (see Appendix A for the whole survey text). We were positively surprised by the number of EPFL students that took the time to answer the questions. At the end, a total number of 653 people took our survey.

4.1 Familiarity with VoIP

The first questions concerned the familiarity with VoIP. We asked to user whether they have already heard something of VoIP and where they got their information from.

4.1.1 General knowledge

The outcome of the first question (depicted in Figure 4.1) was pretty much what we expected: A lot of users know VoIP from friends or from the Internet. This is not further astonishing since the flow of information in the Internet and between friends is extremely high, especially at a technical institute such as EPFL. This also explains the quite significant number of people having heard about VoIP at school or during courses and lectures. Since the market for VoIP products and services is still growing and has not yet reached a mature state, advertisements are not yet very aggressive and therefore, the number of people having heard from VoIP from this information source is quite low.

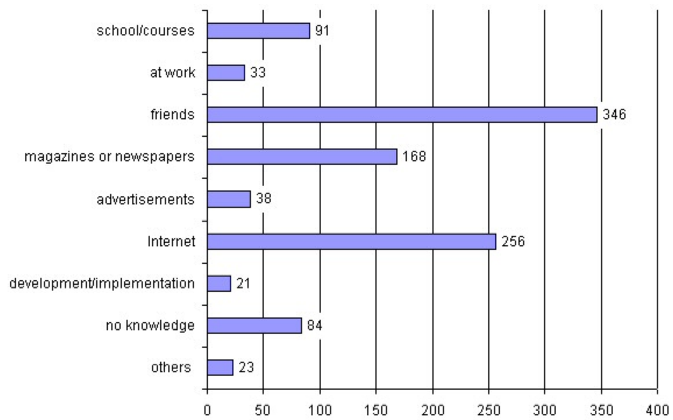

Figure 4.1: Where do you know Voice over IP (VoIP) from?

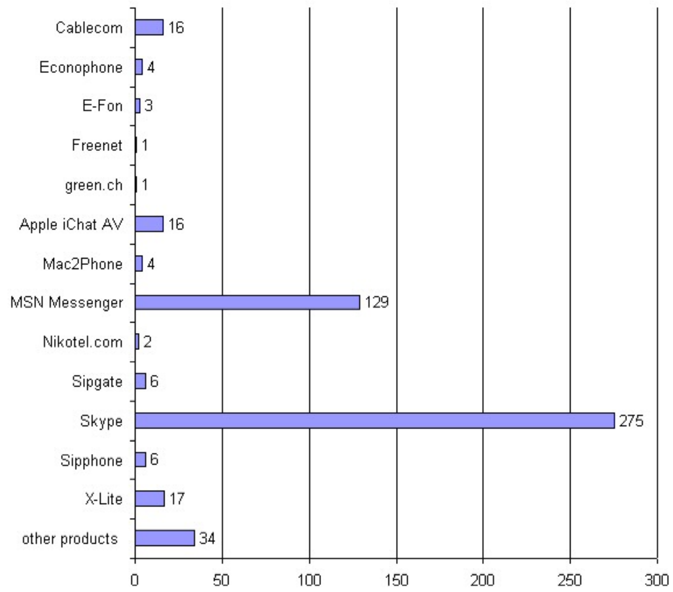

Figure 4.2: Which of the following VoIP services have you already used?

4.1.2 Experience

This part of the survey tried to gather information concerning the use of VoIP products and services. Although VoIP is a relatively new technology, over 50% of the participating users have already used VoIP in any form. We imagined that VoIP might be used only by a smaller fraction of the sample population.

Figure 4.2 shows the result of the next question. Concerning the available services and products, our first assumption was that free services are going to be among the top mentioned since the user does not need to pay anything to satisfy his curiosity. In the end, a large number of people seemed to have already used Skype[1] and the Microsoft Messenger Service (MSN Messenger). The MSN Messenger is already installed on every Windows computer and the preference for this application can be explained by the fact that no additional installation is required. The strong significance for the usage of Skype is clearly due to the fact that the software is available for all major platforms and because installation and configuration is very easy.
However, we were quite astonished that 16 people have already used Econophone's Econostream service. The number of people having used this is equal to the number having already used iChat. At the beginning, we supposed that Mac-users would tend to use iChat, but obviously this is not the case. Maybe, they also switched to the MAC version of Skype.
There were a couple of other products that users have already tried, but no significance could be found for any particular one.

Current usage

VoIP is getting more and more famous (see Figure 4.3) with all kinds of people. Seeing still a predominance for regular fixed line phones, the percentage of users taking advantage of VoIP is constantly growing and the number of people using only a VoIP enabled phone will augment in the future, too.

Satisfaction

Although being linked to more effort the users are in general quite satisfied with the overall quality of the VoIP service they used (see Figure 4.4). Only a few participants put the quality to the lower edge of the spectrum by indicating that they were not at all satisfied. However, this might be due

[1] cf. section 3.3

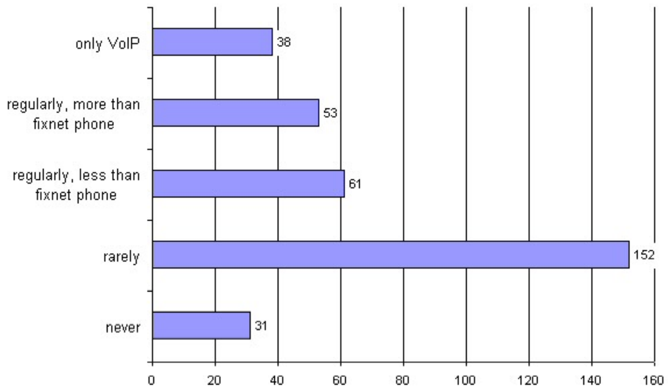

Figure 4.3: How often do you use your VoIP service at the moment?

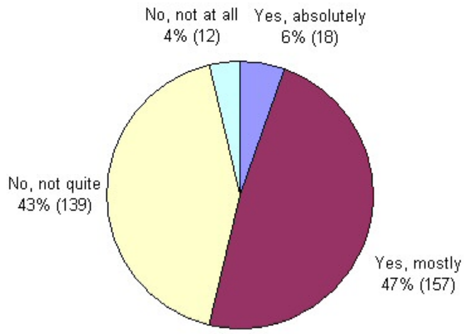

Figure 4.4: Are you satisfied with the overall quality of the VoIP service?

to several facts we did not further investigate in this survey. The key point is that users are mostly happy with what they get.

Usage purpose

Figure 4.5 shows the main usage purposes. For the largest share of users, the cost issue is the triggering fact to switch to VoIP. The ability to make cheap or even free (when using pure VoIP communiation) national and international calls is an important factor. The only disturbing result is that only a very small number of users tend to use VoIP because of the better quality. This is might be a sign that VoIP implementations still have to become better and that the connection quality has to increase.

4.2 Interest and future prospects

The most important part of the survey are the questions linked to the users future intentions (see Figure 4.6). It is of importance for a company to be able to gather an idea of their (possible) customers wishes and needs. The key point was again linked to cost. The chance to make free calls is a crucial need. Furthermore, more technical aspects such as sharing an Internet connection or video telephony are quite requested by the end-users. A few comments concerning the ease of use (e.g. RJ45 plugs for phones etc.) and the security of the communication were also raised.

4.2.1 Expectations

Possible users of VoIP systems have got very high expectations that have to be fulfilled by a provider. The most important need is the ability to call from and to conventional telephones. This is closely related to the requirement of independency from a PC. The second most important requirement is simple handling. User do not want to waste time by reading large manuals and

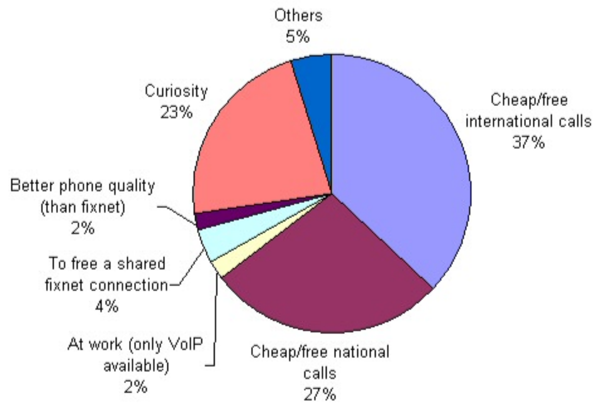

Figure 4.5: Why do/did you use VoIP?

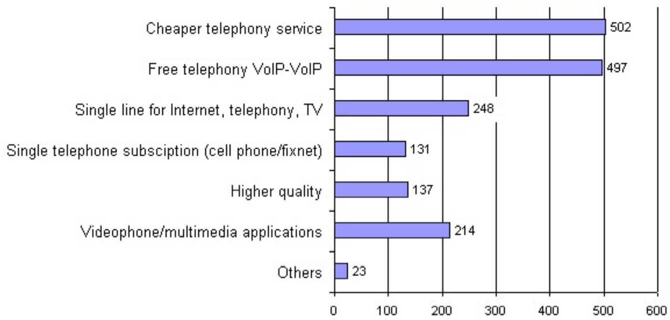

Figure 4.6: What would be the reasons for you to change from fixnet telephony to VoIP?

technical documentation. The installation and use of VoIP products should be straight-forward. The outcome of this question is show on Figure 4.7.

4.2.2 Deterring factors

Another very important aspect of VoIP are the criteria preventing a user from changing to the new technology (see Figure 4.8). The problems named here are crucial and every service provider really needs to care about them.
The most important criterion is the speech quality. Delays, jitter and speech deformation would prevent a great share of the users from switching to VoIP. Temporary interruptions and cut-offs are also to be avoided. Furthermore, the acquisition and maintenance costs for VoIP have to be very carefully looked at.

4.3 Result

The survey clearly showed that VoIP operators have to look carefully at the quality provided and the cost implied by the implementation of VoIP. Although we had quite a large number of people having already used VoIP, this might not be representative for the general population since the EPFL community is fairly technically oriented and younger than the average. Being able to push the new technology to the market needs sincere market studies and analysis.

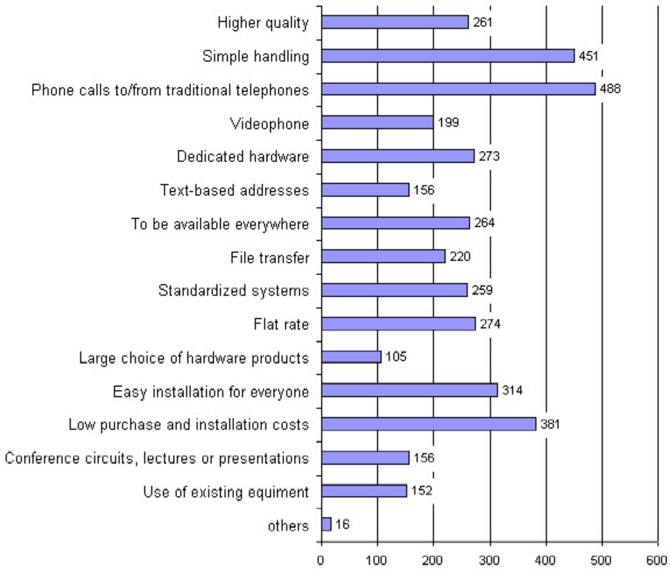

Figure 4.7: What would you expect from a VoIP service?

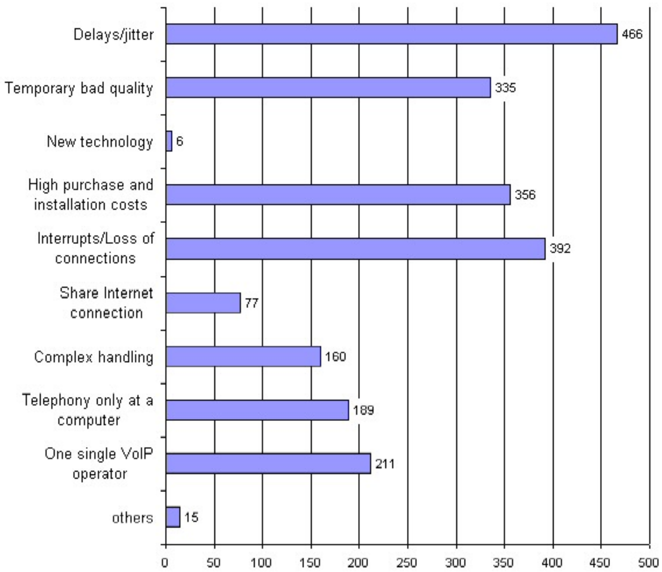

Figure 4.8: What would prevent you mostly from using VoIP services?

Chapter 5

Conclusion and acknowledgements

5.1 Conclusion

This project has shown clearly that an enormous potential for VoIP applications and devices can be exploited in the future. While not being publicly accepted nowadays, there is obviously a lot of interest from the user's and customer's side. Being able to provide cheap, reliable and secure national and international phone calls together with a trustworthy company philosophy and seemless integration into existing infrastructure is the key to success for a VoIP provider. The technology has to be easily available and understandable for everyone in order to play an important role on the market.

We suppose that VoIP is going to dominate the telephony market in the nearer future. This new technology will replace the traditional telephony's tasks and herald a new era of communication systems — enabling flexible telephony combined with video capabilities and secure data exchange. The advantages are obvious and the lack of quality that currently prevents VoIP from being widely used can be overcome by using low-latency and quality-aware networks. Especially in corporate networks where quality of service and data throughput can be assured and controlled, VoIP has already entered a mature state operating at low maintenance and running costs. While investigating the subject we realized that most of the users are willing to provide the necessary acceptance if the service provided is worth it.

For us personally, this STS project has allowed us to gain a deeper insight into the social and economical aspects of a primarily technical issue. VoIP technology will start to enter the mass market. We are now able to understand this process and the related problems and necessities more clearly. Being focused usually heavily on technical aspects, this project helped us to develop an eye for the consequences of technology from another point of view.

5.2 Acknowledgements

We would like to thank the IT responsible of the mentioned company for letting us have a look at their VoIP evaluation documents.

Appendix A

Survey at EPFL

This is the survey conducted at the end of december at EPFL.

Question #1

Where do you know Voice over IP (VoIP) from?
(you may tick multiple boxes)

☐ school/courses

☐ at work, e.g. used as an employee

☐ friends

☐ articles in magazines or newspapers

☐ advertisements

☐ Internet

☐ development/implementation of VoIP software/products (projects, work)

☐ I don't know Voice over IP ...

☐ others

Question #2

Have you ever used a Voice over IP (VoIP) service?

○ yes

○ no (you may continue at question 3)

Question #2.1

Which of the following VoIP services have you already used?
(you may tick multiple boxes)

☐ Cablecom (digital phone)

☐ Econophone (Econostream)

☐ E-Fon

☐ Freenet

☐ green.ch

☐ Apple iChat AV

☐ Mac2Phone

☐ MSN Messenger

☐ Nikotel.com

☐ Sipgate

☐ Skype

☐ Sipphone (MySIPphone)

☐ X-Lite

☐ other products

Question #2.2

How often do you use your VoIP service at the moment (regarding to fixnet telephony)?

○ I exclusively use VoIP (no conventional fixnet)

○ regularly, more than fixnet phone

○ regularly, less than fixnet phone

○ rarely

○ never

Question #2.3

Are you satisfied with the overall quality of the VoIP service you were using the last times?

○ Yes, absolutely, the overall quality is better than telephone (fixnet/cellular phone)!

○ Yes, mostly, the overall quality is often better than telephone!

○ No, not quite, the overall quality is worse than telephone!

○ No, not at all, these overall qualities cannot be compared!

○ Additional comments (What is especially good? What annoys you?):

Question #2.4

Why do/did you use VoIP?
(you may tick multiple boxes)

☐ Cheap/free international calls

☐ Cheap/free national calls

☐ At work (only VoIP available)

☐ To free a shared fixnet connection (flat-sharing community etc.)

☐ Better phone quality (than fixnet)

☐ Curiosity

☐ others

Question #3

What would be the reasons for you to change from fixnet telephony to VoIP?
(you may tick multiple boxes)

☐ Cheaper telephony service (for all calls on mobile and other fixnet)

☐ Free telephony service between VoIP users

☐ One single line for Internet, telephony, TV (e.g. over telephone line, TV, satellite, ...)

☐ One single telephone subsciption (cell phone/fixnet)

☐ Higher quality than conventional fixnet phone calls

☐ Videophone / multimedia applications

☐ others

Question #4

What would you expect from a VoIP service?
(you may tick multiple boxes)

☐ Higher quality than conventional telephone calls

☐ Simple handling (calls, phonebook, ...)

☐ Make and receive phone calls to/from conventional telephones (fixnet and mobiles)

☐ Videophone

☐ Dedicated hardware (telephone sets independent from computers)

☐ Text-based addresses (like e-mail) and no telephone numbers (easy to remember)

☐ To be available everywhere (not fixed), e.g. mobile handset using Wireless Lan/Wi-Max

☐ Integrated data service, e.g. file transfer/exchange

☐ Standardized systems/Full compatibility with all VoIP services

☐ Flat rate (only a subscription fee is paid)

☐ Large choice of hardware products (telephone, video phones, access points, headsets, ...)

☐ Easy installation for everyone (plug and play)

☐ Low purchase and installation costs

☐ Conference circuits, lectures or presentations via VoIP

☐ Use of existing equiment (telephones, fax, ...)

☐ others

Question #5

What would prevent you mostly from using VoIP services?
(you may tick multiple boxes)

☐ Delays/jitter in conversation (e.g. a delay of more than 0.5 sec)

☐ Temporary bad quality (e.g. 2-4

☐ New technology (I do not want to learn something new)

☐ High purchase and installation costs

☐ Interrupts/Loss of connections

☐ Share Internet connection (telephony and data)

☐ Complex handling

☐ Telephony only at a computer

☐ One single VoIP operator (monopoly)

☐ others

Question # 6

If you have further comments or if you are interested in the outcome of this survey (please provide your e-mail address then), feel free to drop us a line.

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Fig 3.1	Swissvoice IP10S. *http://www.swissvoice.net*
	Vtech8100. *http://www.engadget.com*
	Vonage F-1000. *http://www.ehomeupgrade.com*
Fig 3.2	Cisco ATA 186. *http://www.cisco.com/en/US/products*
	AVM Fritz!BoxFon. *http://www.avm.de/en/Produkte*
	(the initial picture is not online anymore)
Fig 3.3	Skype on Linux. *http://www.skype.com/products*
	Xten X-Pro on Pocket PC. *http://www.xten.com/index.php?menu=products*
	Apple iChat AV. *http://www.apple.com/ichat*
Table 3.2	Cablecom "digital phone". *http://www.cablecom.ch/en/digitalphone*