Voice over IP and the Modern Web

Vladislav Lazarov
lazarov@in.tum.de

20.12.2006

Building Next Generation Internet Services for Fun and Profit

Winter Term 2006/07

Supervisors: Dean Jacobs, Ph.D.; Dipl.-Inf. Stefan Aulbach
Chair of Computer Science III: Database Systems,
Prof. Alfons Kemper, Ph. D.
Faculty of Computer Science
Technical University of Munich

Abstract

This is a final report for the topic "Voice over IP and the Modern Web" introduced in the Advanced Seminar "Building Next Generation Internet Services for Fun and Profit" at the Technical University of Munich. It first describes the history of telephony and the problems behind the technology which lead us to Voice over IP. This final report introduces the terminology of Voice over IP and presents then a short overview of the most important existing services and clients. Then it discusses which communication problems are being solved with VoIP, which problems emerge and what are the pros and cons of the technology. Finally, new approaches, crucial issues and the future of Voice over IP and its relations to Web 2.0 are being discussed.

Contents

Introduction

Since its invention the telephone has never been playing such an important role as now. It is impossible to think of a successful business which can live without the use of voice communication. The introduction of the world-wide web a few years ago did bring to us many new key technologies. Originally, e-mail was supposed to be the killer application of the Internet. Today, instant messaging and VoIP are the new state of the art communication technologies. Voice over IP is bringing new successful business models for new Internet start-ups and is changing the market space like never before. The old telco companies are now struggling to catch the new vibe, while many new entrants begin rolling out commercial VoIP services. In the same time policymakers around the world are grappling with the regulatory implications. This particullarly applies for the two largest near-term VoIP markets: the United States of America and the European Union, but also to the fastest growing technology market Asia. The world today is moving towards a unified network which will bring together all needed for business and fun technologies to go through a single wire. Thus, leading to telco companies entering not known before areas such as providing television and video on demand, while other Internet providers, such as cable televisions are moving into the communication business.

1 Historical Overview

When trying to find the roots of Voice over IP we have to start our voyage from the invention of the telephone. Back in 1874 the telegraph was the fastest way of delivering messages to people. On March, 7th 1874 this was to be changed. On this date the US Patent office issued patent 174,465 to Alexander Graham Bell which covered “the method of, and apparatus for, transmitting vocal or other sounds telegraphically by causing electrical undulations, similar in form to the vibrations of the air accompanying the said vocal or other sound”, the telephone. Since then the voice communications became an essential part of ones life. We can distinguish three major “waves” in the telecommunication business.

1.1 Voice as an Analog Signal

The first wave is the one which allowed voice to be carried as an analog signal through the network. One of the main drawbacks behind this is the problem that noise was increasing with the distance. This makes long-distance calls a difficult experience to both sides involved in the conversation. The other big problem was the increase of the demand of communications especially after World War II.

1.2 The Digital Communication Revolution

The early 1950s introduced a technology that converted speech into digital signals which allowed the voice to be coded in 0's and 1's and transferred through the network until it reached the other side, where it was again decoded into analog voice. This solved the problems of analog communications, because the impact of distances was decreased with the usage of digital repeaters, which regenarated the lost 0's and 1's and eliminated some of the noise. Furthermore, other noise-elimination techniques were also introduced. Since one single T1 line could carry 1.544 Mbit/s, while the European equivallent an E1 line was able to carry 2.048 Mbit/s the problem with the increasing demand of communication and the capability of the network to deal with it was also solved.One key technique was also introduced to be used with digital signal communications - TDM. The term stands for Time Division Multiplexing and is described as combining numerous signals for transmission on a single communication line or channel.

Both approaches allowed a single call to have a reserved badwith of 64 kbit/s. The connection is established during the dialing proccess and is then being held for the duration of the call and after that it is closed. In this way the signal is going through a single reserved for that call route.

1.3 Advantages of Traditional Phones

- Since the networks follow a good and long-time established model they can offer really good reliabilty. Most of the phone companies are known to offer the so called five 9s, which means around 99,999% uptime a year. This is less than 5 hours of downtime/year.

- 64kbit/s are devoted to a single call. Therefore, there is almost perfect sound when calling somebody using the PSTN network. Mobile phones use a bandwith of 10-15kbit/s, which is the reason that sometimes there are odd sound effects.

1.4 Problems with Traditional Phones

There are still crucial issues for the traditional phone network which have to be solved.

- It is still very hard to implement new services and features. In order to include a new service the phone companies have to reprogram the complex switches, which is not really a trivial task.
- The usage of unnecessary bandwidth leads to higher cost of long-distance calls.

1.5 The Third Wave

As in the previous example previously mentioned problems are being solved in the so called "third wave" of voice communications which is the Internet Telephony, while new drawbacks and challenges emerge. The proccess is involving a major change. The currently circuit-switched networks are moving into the packet-switched ones. Hereby, voice packets are not using a single line during the call. Packets can take different routes to reach the communicating partner. What is more, only the necessary bandwidth is being used and no packets are being sent when there is silence, for instance.

One of the key changes is the one of the communicating units. When telephony is being done the traditional way end-users have access to pretty dumb devices, which send data through a sophisticated network where smart machines (the so called switches) decide how to route the signal. The situation in the packet-switched networks approach is different. Here the end-user has access to more sophisticated machines (computers or smart-phones) and the data is sent through a dumb network, which involves a variety of cables, wires and routers.

2 Internet Telephony

2.1 Definiton

Internet Telephony is defined as the process of routing of voice conversations through the Internet or any other IP-based network.

2.2 The Technology

Since Internet Telephony is involving two different research topics - the Internet and the telephone there are also two technological approaches.

The International Telecommunication Union (ITU) has introduced the H.323 standard [Rec00]. It is an "umbrella" specification proposed at first in 1996 and has been updated several times. It suggests a standard for multimedia communication over packet switched network such as LANs, WANs and

Internet. The Session Initiation Protocol (SIP) [JR02] was introduced by the Internet Engineering Task Force. It includes a suit of call setup and media mapping protocols for multimedia communication over the packet-switched network. Due to its simplicity, flexibility and its modular character SIP is becoming one of the most promising VoIP protocols which has attracted many VoIP vendors to provide SIP based VoIP solutions.

From the telephony communication history point of view, the PSTN can be considered as the traditional communication method, whereas comparing with SIP H.323 can be considered as "legacy" VoIP protocol, although H.323 is still widely used. The roots of SIP as a method-based protocol can be found in HTTP, while H.323 is following the established PSTN model with a huge protocol stack.

2.3 VoIP in Numbers

During the past few years VoIP has established as a fast-growing technology. As of December 1st, 2006 there are 286 909 716 downloads of Skype (one of the leading VoIP "soft-phones" and service providers at the moment). There are more than 100 000 000 registered users of Skype. Excluding this numbers we can take a look at the business in the United States [Che05]. For the year 2005 there are more than 400 VoIP service providers. Around 1 Mil. out of 106 Mil. households are using VoIP. 50% of that household market is being held by Vonage and Cablevision, which makes them the two major players. The annual revenue only in the USA for 2005 is $2,5 Bil. It is estimated [Che05] that in 2009 12 Mil. households will be using VoIP instead of conventional telephony and the revenue will be $21 Bil.

2.4 Incumbent to Watch VoIP Services

There are a few major players in the market sphere of Voice over IP.

2.4.1 Skype

Skype is the best-known world-wide Voice over IP service. It is a proprietary VoIP system developed by Skype Technologies S.A., which was founded in 2003 by Niklas Zennstroem and Janus Friis. Garfinkel [Gar05] concludes that Skype is related to KaZaa. Both companies were founded by the same individuals, there is an overlap of technical staff, and much of the technology originally developed for KaZaa is used in Skype. Therefore Skype is using a peer-to-peer approach for the connection between individual users.

Skype offers three services: VoIP allows two Skype users to establish two-way audio streams with each other and supports conferences of up to 4 users, IM allows two or more Skype users to exchange small text messages in real-time, and file-transfer allows a Skype user to send a file to another Skype user (if the recipient agrees). Skype also offers paid services that allow users to initiate and receive calls via regular telephone numbers through VoIP-PSTN gateways.

The business model of Skype includes also other paid services such as Voicemail, SMS, Call forwarding, Ringtones, etc. There are versions of the soft phone for Windows, Mac and Linux. Moreover, there are also versions targeting different user groups, which offer more or less the same features, but the most used for the specific group are optimized for them. Therefore, there is a version called "Skype for Business". The company is offering also a version optimized for portable devices such as PDA's or Smart Phones called "Skype for mobile devices".

Although Skype is not open-source and does not offer an API for other developers in order to integrate the service in their own web services, there are many community-driven or community-created extras available on Skype's website.

To increase the number of North-American users Skype offered a free calls plan, which offered unlimited and free calls to landline and mobile phones within the United States and Canada. As of January 1st, 2006 the company will offer a flatrate for unlimited calls for an annual fee of $30 a year. [Ric06]

In the fourth quarter of 2005 online auction company Ebay acquired Skype for more than $2.6 billion. [NB05]

2.4.2 Gizmo Project

The Gizmo Project is created by the team who built the SIPphone VoIP platform. SIPphone builds and operates all the servers and services on which hardware, software or web community can offer free VoIP services. At the core of Gizmo Project is a commitment to open standards, which is critical to deliver the true potential of VoIP. SIPphone believes that like web pages, email and IM, calls should be free.

The team behind Gizmo Project has their roots in some of the Internet's most influential start-ups, including MP3.com (founder), i-drive (founder), Linspire (founder), Scour (founder), and Napster. The founder of Gizmo Project's parent company is Michael Robertson, the influential entrepreneur behind MP3.com.

Like Skype Gizmo is also using a peer-to-peer model for connecting users. Unlike Skype, it is based only on the SIP protocol for calls and is using the

Extensible Messaging and Presence Protocol (XMPP) for instant messaging, thus allowing it to connect with other services based on SIP (OpenWengo) or XMPP (Jabber Clients, Google Talk).

Just like Skype it is offering free PC-to-PC calls and has a CallOut feature allowing the connectivity to traditional PSTN networks.

Over 100 organizations have peered their networks with the SIPphone network, so users can exchange calls for free. Many of the university networks in the USA are peered in the project including Univ of California, San Diego; Univ of California, Santa Cruz; Univ of California, Irvine; Univ of the Philippines and more.

One of the most important milestones of the Gizmo Project was the announcement of its "All calls free" plan. It applies when both call participants are active Gizmo Project users making a few phone calls per week with Gizmo Project. Free calls may originate from anywhere in the world, but must be to a qualifying number in one of the 60 countries for which the plan is offered. Calls must be made from the caller's contact list to either the "home phone" or "mobile phone" number the call recipient included in his or her profile, and both parties must have shared each others profiles with one another.

Furthermore, Gizmo is offering also access numbers, which allow a Gizmo user to receive a call from the traditional phone system directly to the users soft phone. Currently there are 20 access numbers in the USA.

2.4.3 Google Talk

Google Talk is part of Google's strategy to organize the Internet. Based on the SIP protocol it is providing registered Google users the opportunity to make free PC-to-PC calls. It is also integrating Google's e-mail, instant messaging and call service.

In August 2006 Google announced a multi-year agreement to connect users, merchants, and advertisers around the globe they signed with Ebay. [Rel06] This could possibly lead to Google Talk and Skype's interconnectivity.

2.4.4 Vonage

Unlike the previously presented services Vonage is not a peer-to-peer network. More like a traditional phone company Vonage is redirecting all calls through their own servers. Vonage is described as a new-wave phone company in the USA and Canada, offering an all-inclusive phone service using existing Internet connection. It includes free calls to Europe landline phones and other interesting features such as Voicemail, Caller id, 911 Dialing, 3-Way calling, Area Code selection and Click-to-call service.

2.4.5 Asterisk PBX

Moving further, there are other VoIP start-ups that are interesting to watch. One of them is the Asterisk PBX (private branch exchange) system. Asterisk is an open source telephone system. It is offering flexibility, functionality and features not available in advanced, high-end (high-cost) proprietary business systems. Asterisk is a complete IP PBX for businesses, and can be downloaded for free.

The key idea behind Asterisk is providing PBX as a software. It runs on a wide variety of operating systems including Linux, Mac OS X, OpenBSD, FreeBSD and Sun Solaris. It is integrating almost all protocols and can interoperate with almost all standards-based telephony equipment.

Asterisk has made one of the most important steps for integrating Internet Telephony in businesses, allowing them to have not only cheap or free calls, but also a relatively cheap PBX system.

2.4.6 Iotum

Based on Asterisk PBX Iotum is another VoIP startup. It is a Web 2.0 call management application that utilizes the user's context information - who is calling and what the user is currently doing - and preferences to know which calls are important and which can wait. It is simple for users to configure and seamlessly connect to their calendar and instant messaging (IM) tools - making filtering and routing decisions based on who is calling, time of day, what is on the calendar, location, etc.

Like a search engine, Iotum's Contextualizer constantly indexes and categorizes contextual information. It provides the "what, where, when" information to the Rules Processor, which determines if and how a call should be handled. Using an easily implemented XML-RPC Services Interface, Iotum connects to a Softswitch, Media Gateway or IP-PBX. [iot06]

2.5 The Business Models

One of the major questions when talking about Voice over IP is: "Who (if anyone) gets paid for VoIP?". Most of the free PC-to-PC call services do not have any expense at all for providing the service since they are using a peer-to-peer approach. The real question then is how is the low-cost of calls achieved when connecting to the traditional PSTN network.

The VoIP service providers' real expense is only for the so called "last mile". The solution which VoIP providers are using is paying local phone companies to ring the phone and complete the call. Therefore the same rate can be offered for everyone, no matter if the person who is being called lives next door or 5000 km away.

The VoIP providers buy prepaid minute packages from local phone companies which have to be used over a period of time and then redistribute them to the users of their own system. At a wholesale level one minute costs around 1 ct.

There are different approaches for the redistribution of the minute packages to the users. Like mentioned above there can be “flatrates” which include unlimited calls over a period of time (a month with a montly fee, a year with an annual fee, etc.) or minute packages. RNK Telecom offered an interesting service which offered lifetime free VoIP calls to PSTN networks for a one time fee of $999.

Since in some countries local telephone companies are feeling threatened by the increasing usage of Voice over IP they are no longer dealing with VoIP providers. A good example is Romania, where traditional phone companies do not accept the offers of Skype. Therefore Skype calls to Romania are not cheaper than calling the country from ones local telephone company.

3 Crucial Issues for Voice over IP

There are still many things that have to be done in order to have pure Voice over IP communications. Since implementing new features is easier than reprogramming the complex PSTN switches, we still can categorize Voice over IP as a disruptive technology.

3.1 Quality of Service

One of the key concerns for businesses is the Quality of Service for a phone call. Voice quality is not the same as this of traditional phones. The roots of this problem can be found in the IP Topology.

There are three main worries:

- **Latency** - because of a network congestion packets can be delivered too slowly. A delay of more than 150ms will lead to a strain of natural communication.
- **Jitter** - the arrival time of packets may be different. Since packets can take different routes through the network some packets can arrive not in the correct order.

- **Packet loss** - packets can be dropped, or simply not arrive. Generally, voice codecs are tolerant to packet loss, but a loss more than 2% to 5% has a serious effect on quality.

All mentioned points can lead to odd sound effects, echoes or silences during a call. There are already efforts to guarantee a good Quality of Service. One of the solutions includes implementing a jitter buffer, which has to rearrange the packets properly.

Other proposals tend to guarantee minimum bandwith using the Resource Reservation Protocol (RSVP). This is a protocol that supports the reservation of resources across an IP network. Applications running on IP end systems can use RSVP to indicate to other nodes the nature (bandwidth, jitter, maximum burst, etc.) of the packet streams they want to receive. RSVP depends on IPv6. Also known as Resource Reservation Setup Protocol.

Another interesing approach is MPLS. This is a scheme typically used to enhance an IP network. Routers on the incoming edge of the MPLS network add an "MPLS label" to the top of each packet. This label is based on some criteria (e.g. destination IP address) and is then used to steer it through the subsequent routers. The routers on the outgoing edge strip it off before final delivery of the original packet.

A solution can also be implementing a so-called "Type of Service" (TOS) label, which can be used to give priority of certain packets. The main problem will then be the principle of network neutrality.

3.2 Reliability

While traditional phone companies can offer the mentioned above five 9s, VoIP provides usually do not guarantee even three 9s (8h downtime/year). Moreover, a crucial worry is what happens in the event of a power outage. While traditional phones can function without power, VoIP must have more sophisticated devices which usually need power in order to function.

Nevertheless, reliability of VoIP, cannot be a serious problem nowadays since there are enough solutions for this. One of the interesting facts is that on 9/11 VoIP "survived" the crash, but there was a destroyed central plant of "Verizon", which lead to a total outage of traditional voice communications.

3.3 Emergency Calls

When there is an emergency call nowadays traditional phone companies send all the needed information to the emergency personal, such as the location from which the call was attempted, which allows them to react immediately.

This still has to be implemented for instance on the basis of the IP adress in VoIP providers. Since there are still many Proxy services and NAT-traversals it will be still hard to really locate the caller or this will take more time.

Another key question when using a VoIP service and calling an emergency short number such as 112 in Europe or 911 in the USA is where to forward the call. While it is easy to forward the call when using landlines (based on the telephone location) or mobile phones (using the cell in which the user currently is) it is still hard to do this with the IP address.

3.4 Connectivity With Other Services

The main goal of Voice over IP has to be “maintaining a unified communications network including all necessary services for business and for fun”. Nevertheless, until now there is no real connectivity between all the VoIP service providers. It is not stated neither in European, nor in American laws, that VoIP providers should offer connectivity between each other. Moreover, most applications do not offer APIs making it hard to impossible to create your own services based on them. Another thing that has to be mentioned is that Voice over IP is still not part of the global telephone system.

3.5 Security

Many people fear the Internet as potentially insecure for transmitting private data. Therefore, they do not trust Voice over IP as a serious and secure mean of communication. The main thing behind these worries is the IP network topology and the possibility to intercept packages. But actually, VoIP is even more secure than the traditional phones. Even the free services such as Skype offer crypthography, which makes it probabilistically impossible to hear the phone call.

3.6 Network neutrality

Tim-Berners Lee defines network neutrality using this sentence: “If I pay to connect to the net with a given quality of service, and you pay to connect to the net with the same or higher quality of service, then you and I can communicate across the net, with that quality of service” [Lee06]

The worry with network neutrality and Voice over IP is that traditional phone companies are feeling threatened by the increasing popularity of VoIP, which leads to less income for the telco’s. They now have to think of a way to survive in the new setting. In the USA there were already attempts to break the network neutrality law which would have lead to unequal competition, since big companies would pay for their service to be prioritized and would disallow their opponents.

Another issue is the possibility of filtering content, how it is still done in China and other countries lead by totalitarian regimes. Voice over IP is currently banned in UAE.

3.7 Regulations

There are still no regulations of Voice over IP. Regulations could solve many of the previously mentioned problems such as emergency calls, connectivity and the breaking of the network neutrality concept. Moreover, the techniques used by some companies are also leading to problems. American university networks are beginning to ban Skype [Pau06] because of its "supernode" functionality and the fact that there is no way to turn off this option. Skype is using a "grid-computing"-like scheme known as "supernode" functionality which allows calls or file-transmissions to be relayed through the users' computer. Skype's Terms of Service include "general usage rights" of the network, that might not belong to the Skype user.

Other things that have to be regulated are the usage of private data and possible records of calls that can be made.

4 New Approaches in VoIP

Voice over IP offers many different ways to present new Internet start-ups based on it. There are numerous business models and opportunities. This section is presenting some of the possibilities for the future of Voice over IP in the Modern Web.

4.1 VoIP and Web 2.0

Web 2.0 was first coined by the O'Reilly Media in 2004. It refers to a perceived or proposed second generation of Internet-based services like social networking, wikis, communication tools. All can be summarized by the need of one to collaborate with other people. Voice communications are an essential part of the social life and therefore voice combined with the Internet can have a huge impact on Web 2.0. Currently many websites, especially web shops or product vendors offer a "click-to-call" opportunity allowing direct voice interaction of the user with the website staff.

Recent publications suggest [Mal06] that Adobe, who recently bought Macromedia are planning to offer techniques to include Voice over IP into

Flash, which will offer many new opportunities of user interaction without the need of a soft phone, such as pure hosted services for VoIP and even more.

There are already approaches for embedding Voice over IP into websites. One of the biggest blog providers Livejournal now offers the opportunity for their users to talk with users of the Gizmo (and with it the whole SIPPhone) network.

Voice over IP is more and more embedded in games. Multiplayer games today have the biggest revenue in the game market. What is more MMOGs (Massively Multiplayer Online Games), which have become popular since several years and have now spawned a whole subculture, are now implementing VoIP embedded services. Really popular Web 2.0 games such as "Second Life" are now adding VoIP functionality. What is more in October Vivox announced their "one million free minutes" promotion which enables the 3D virtual world "Second Life" users to speak to each other via their phones using PC VoIP to landline termination (PC-to-PSTN). Natively Second Life doesn't support PC-to-PSTN, however users have been able to use other third-party workarounds, like TeamSpeak. Vivox has an interesting concept whereby they are building "phone booths" into the Second Life virtual world that allow residents to call out to any telephones from them. They are also building microphones into the virtual world that will allow users access to voice chat of up to five people at a time. [Kea06]

Moreover, directories based on VoIP contact sharing can be build. There are already approaches for this implemented by Google or the popular OpenBC service.

4.2 Real-time Multimedia Communication

Providing Internet Telephony allows integration of existing online communication channels, and therefore a more sophisticated calling experience. Instant Messaging can be used for presence handling/availability information. This can solve the problems, which occur if the called person is unavailable and can reduce time cost.

Moreover, Voice over IP can evolve to a Desktop Sharing Application using real-time multimedia (audiovisual presentation). This is particularly useful for education (eLearning) or for virtual business meetings.

4.3 Semantic Web

Voice over IP can also profit from Semantic Web techniques. The already existing XML-based APIs can be used to access presence or availability information. This can be used in connection with automatical initiation, acceptance and redirection of calls (useful for example for call centres). The automatic proccessing of that information can be used for offering many useful services to users.

4.4 Wireless Internet Telephony

The first and most important steps to wireless Internet telephony are already done. With the creation of optimized soft-phones for mobile devices and the big amount wireless hotspots it is just a matter of time that VoIP can replace not only landlines, but also mobile phones. The real paradox today was mentioned by Henrik Levkowetz (a software developer ipUnplugged) [Che05]: "Today communications can offer seamless handover between cellular and WiFi, but not between WiFi hotspots" One of the current research topics is, therefore, how to move from one ISPs hotspot to another and how to transfer the session.

5 Summary and Conclusions

As stated before we can for sure classify Voice over IP as a disruptive technology. It is just a matter of time for it to replace our phones. Voice over IP will lead us to less complexity and to maintainance of one single network. It is opening the doors for many new and exciting services for Internet users. Moreover, traditional phone companies will have to change in order to continue existing. We are moving towards a state where one single vendor will bring to us any necessary means of communication for our free time or business. Hosted services will replace todays softphones, which will allow us to be reached whenever and wherever we want. Data and voice will be provided over one single unified network, which will allow full integration of communication into multimedia. In order this to be done there are many implementation challenges. Laws and regulations still have to be discussed.

References

[Che05] Steven Cherry. Seven myths about voice over ip. *IEEE Spectrum*, 2005.

[Gar05] S. L. Garfinkel. Voip and skype security. January 2005.

[iot06] iotum.com. Our solution, 2006.

[JR02] G. Camarillo A. Johnston J. Peterson R. Sparks M. Handley E.Schooler J. Rosenberg, H. Schulzrinne. Sip: Session initiation protocol. *RFC 3261, IETF*, June 2002.

[Kea06] Tom Keating. Voip in second life. *TCMNet: VoIP and Gadgets Blog*, October 2006.

[Lee06] Tim-Berners Lee. Net neutrality: This is serious, June 2006.

[Mal06] Om Malik. Flash in the voip pan. *GigaOM*, September 2006.

[NB05] Terence Neilan and Ken Belson. Ebay to buy internet phone firm for $2.6 billion. *The New York Times*, September 2005.

[Pau06] Ryan Paul. More universities banning skype. *Ars Technica*, September 2006.

[Rec00] ITU-T Recommendation. H.323: Packet-based multimedia communications system, November 2000.

[Rel06] Google Press Release. Google and ebay sign multi-year agreement to connect users, merchants, and advertisers around the globe. August 2006.

[Ric06] Matt Richtel. Skype's free phone call plan will soon have annual fee. *The New York Times*, December 2006.