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VoIP - Voice over Internet Protocol

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1 Abstract

Over the last years VoIP has become a buzzword, not only in computer related environments. The abbreviation VoIP means Voice over Internet Protocol and describes a technology of transferring voice and control information (in the form of digital data) over a standardized IP based network like the internet. Because of the cost effectiveness, VoIP has totally replaced the Plain Old Telephone System (POTS) in some companies. Private use of VoIP has become a boom, last but not least through free software based telephones like Skype[1]. Those soft phones[2], small pieces of software which make the use of VoIP as easy as using a web browser, can be installed on nearly every modern computer. On the other side plain old telephones can be connected to VoIP networks using special adapters.

This Article describes the technology around VoIP, the applications and the propagation of VoIP.

[1] http://www.skype.com/
[2] http://www.google.de/search?hl=de&q=define:softphone

2 History of Communication

Since the first establishment of a telephone line in the late 1870s communication has changed in a multitude of ways. First only local calls within the small private networks for example within Berlin were possible. Soon bigger telephone networks were built and connected to even larger networks. In 1900 calls could be made from Berlin to Paris. In Germany this Plain Old Telephone System remained unchanged as far as possible for around 80 years.[1]

The first really important change to the telephone network in Germany was the introduction of the digital ISDN (Integrated Services Digital Network), which started in the 1980s and was finished by 1995.

Local Area and Wide Area Networks (LANs / WANs) have been developed independently from the telephone networks. Phone lines, which were publicly available in nearly every household, where used to connect to those networks. Today the telephone and "data" networks are conneted to each other. Data can be transported over the phone network and calls can be made through computer networks. The key to connect those networks is a protocol developed in the late 1970s: the Internet Protocol (IP).

[1] http://de.wikipedia.org/wiki/Geschichte_des_Telefonnetzes

3 Voice over Internet Protocol

Voice-over-IP, also called IP-telephony, is sometimes described as a technology for making calls over the internet. Technically it means the transport of voice, encoded as digital data, over IP based networks like the internet or a company's intranet. This connection has not to rely entirely on an IP based network. Part of this connection can also be transported, through specific gateways, over a usual landline.

The first brick for VoIP was laid by the company VocalTec[1] when they released their telephony software "Internet Phone". Everything that was needed was a Personal Computer with a soundcard connected to a network. Since the quality was rather bad compared to the quality of the Plain Old Telephone System, a commercial use wasn't really possible.

In 1996, a new protocol called SIP (Session Initiation Protocol) was introduced. This protocol increased the quality of the calls and made audio conferences (the connection of more than two conversational partners) possible. Before this standardized protocol the conversational partners usually even had to use the same software to talk to each other. Devices or soft phones which support SIP can communicate with each other independent from the vendor now.

Over the years the costs of bandwidth went down, PCs got more powerful and the

[1] http://www.vocaltec.com/

codecs, which are used to transform the voice into digital data, became more advanced. VoIP today is in a status where it could replace the plain old phone in matter of costs and speech quality.

3.1 How does VoIP work?

Voice is recorded as analogue signal through a microphone in the same way as in the Plain Old Telephone System. To transform this analogue signal into digital data codecs are needed. This digital data is then fragmented into smaller "packets" which are sent over the network to the recipient.

There are lots of different codecs with their respective advantages and disadvantages.Depending on the bandwidth available on the network a codec which produces a signal of lesser quality, which in turn means a smaller amount of data, may be used. If a higher bandwidth is available a codec producing a better signal with a higher quality might be used.

On the other side, the digital data stream is reassembled from the received packets and decoded into an audio signal which is then replayed on headphones or speakers.

3.2 Setup

VoIP calls can be made basically in three different ways.

The easiest and cheapest way to use VoIP is to connect a simple headset to a computer and use a VoIP software like Skype[2], Gizmo Project[3], Google Talk[4] or VoIP Buster[5]. Those soft phones simulate a telephone in the form of a small piece of software. This option needs minutes to set-up and is very cheap. Everything needed for starting with making VoIP calls is a simple headset and a client software (which can be downloaded from the internet mostly free of charge). This option is also perfect for people travelling a lot, because they usually do not even need a headset. Most laptops these days come with a built-in microphone and speakers. Regarding privacy, using headphones might be appropriate never the less. One can be reached under the same number overall in the world. However there is one big disadvantage in using this method: The computer has to be switched on and connected to the internet in order to make or receive calls.

The second and most convenient way, once setup, is using the same standalone phone used for landline calls. Over a small adapter (an analogue-to-digital converter) the telephone can be connected to the LAN. This adapter contains all hard- and software necessary to convert the analogue voice signals from the phone to digital IP Packages, which then can be transferred over a private network or the internet. The use of VoIP in that case is nearly transparent. There are numerous kinds of converters. The big advantage in this case is that the computer can be switched off and it is still possible to receive and initiate calls over the internet. Nowadays these adapters sometimes come built-in in internet routers. The company AVM[6] for example offers their Fritzbox! Fon router with the option of analogue ports to connect two plain old phones and use them for VoIP. Since they also sponsor an input for a connection to the Plain Old Telephone System a user can define which calls are made over VoIP (for example long distance calls) and which calls are made over the Plain Old Telephone System (for example local

[2]http://www.skype.com
[3]http://www.gizmoproject.com
[4]http://www.google.com/talk
[5]http://www.voipbuster.com
[6]http://www.avm.de/en/index.php3

calls).

When it comes to commercial use, these analogue-to-digital adapters do not perform well. For every single phone, one of these adapters is needed. The maintenance of a telephone system built up with those converters is nearly impossible. There are special telephones, which can be connected to the network directly. These telephones or IP-phones have all the software needed to use VoIP built-in. In large company environments, there are also a lot of commercial systems (as well as free ones) available to set-up large scale telephony environments. Especially the open source software asterisk[7], which is a full featured private branch exchange (PBX) in form of software, can help companies to reduce their costs for telephone equipment. The replacement of some private branch exchanges is in some cases more expensive than the purchase of a whole VoIP infrastructure including the setup of an asterisk or commercial server. These IP-Phones can also be used without a complete VoIP infrastructure. Soon there will be models available which communicate with the network over Wireless LAN, those phones can then replace the usual cordless phones.

Why should a company replace their existing telephone infrastructure?

The main reason could be cost-reduction in the field of long distance calls. If long distance phone calls are routed over the internet, the costs are nearly zero. The provider of the gateway connecting the call to the distant POTS usually charges less than for a local call.

If a company has departments in different locations they usually have a network link between those departments to interchange data. That connection could also be used for transporting the phone calls within these departments at virtually no cost. (Assuming the bandwidth is sufficient enough.)

[7]http://www.asterisk.org

3.3 Problems / Risks

Like nearly all other great inventions VoIP comes not without any difficulties or problems. One major problem is that, in case of an emergency, the location of the caller can not be determined as easy as in a private run telephone network. It might even not be possible to call the local emergency hotline since even the region prefix might not be known to the provider.

Another big issue is the security of the calls made through a VoIP network. Since the call, encoded as digital data, is transported over different computers a possible attacker could just record these data and reconstruct the phone call. There are already programmes available automating this process. But in reality it is not as easy as it sounds. Larger companies can set-up secure networks within the internet or the intranet connecting different departments. Connections within these virtual private networks (VPN) are invisible to the outside and hence harder to compromise.

Since VoIP is built on top of an IP based network these networks have to be maintained and have to offer enough bandwidth to let VoIP pass through with the expected quality. If VoIP is used over the internet, which at least for private and small business users is mostly the case, this bandwidth and thus the quality is not guaranteed because they have no influence on the traffic on the internet. There is also no one who can give support for the internet because it belongs to no one specific.

3.4 Future Developments

Besides the soft phones the most used phones for VoIP are the IP Phones. These phones, with their built-in network interface and software, are able to connect to a LAN and therwith to a VoIP service directly.

Another killer application would be the use of VoIP while on the road. Google as well as other companies discovered this and there will be phones available in the future which could be connected to a VoIP gateway over a publicly available Wireless LAN. In case there is no wireless network available, these phones could connect to a GSM network instead. One of those phones for example is produced by alpha networks, which will be also able to connect to Google's proprietary VoIP Network: Google Talk[8]

[8]http://peripherals.engadget.com/2006/08/03/alpha-networks-is-prepping-google-talk-wifi-phone