VOICE OVER IP AND NETWORK CONVERGENCE

Assaid Othman Sharoun
Doctoral Degree Programme (1), FEEC BUT
E-mail: xsharo00@stud.feec.vutbr.cz

Supervised by: Vladislav Musil
E-mail: musil@feec.vutbr.cz

ABSTRACT

As the IP network was primarily designed to carry data, it does not provide real-time guarantees but only provides best effort service, which is inadequate for voice communication. Upper layer protocols were designed to provide such guarantees. Further, as there are several vendors in the market implementing these protocols, conformance to standards and interoperability issues have become important with general VoIP (Voice over Internet Protocol) description noting VoIP advantages, disadvantages, problems with reliability and security. There are also mentioned commonly used VoIP protocols (H.323, SIP and IAX), characteristics of code's and solutions for quality of service. Last part is concerned with practical realization of the integrated PBX (Private Branch Exchange) consists of POTS (Proprietary Old Telephone System) connection which is realized by the ISDN (Integrated Services Digital Network) interface, old ISDN PBX Fritz!X and VoIP phone connected over Internet.

1 INTRODUCTION

With IP telephony is coming new age of the voice communication. This transformation had begun at the time when the requirements of the data communication exceeded the possibilities of ordinary public switched networks. Telephone lines were used for data transfers at early stages but current data transfer rates are much higher than those used for telephones. For example data rate used by only one speech call takes 64kbit/s but common WAN speeds offered by cable networks reaches ranks of Mbit/s at both up-link and down-link. The ability to transfer voice over the Internet, rather than PSTN was first made possible in 1995 when Vocaltech company released its Internet Phone software. This software was usable on a multi medial personal computer equipped with sound card, microphone, speakers and modem. Internet Phone was designated to run on a machine with at least 486/33 MHz processor. This program converted the voice signal into IP packs that were transmitted over the Internet. The complication of this VoIP solution was that it worked only at the time, when both parties were using the Internet Phone software and both parties had to be on-line at the same time.

2 VOIP

2.1 THE ADVANTAGES OF VOIP

VoIP can offer some potential advantages and benefits to its user:

1. Lower charges 2 .integrated services and greater user features
3. Flexibility 4. reduced infrastructure

The most important advantage of VoIP are implied cost savings, but we can not expect savings immediately due to the upgrading and implementation costs. Phone calls made by the VoIP are free or costs less than equivalent services provided by the traditional telephone operators but are little bit similar to the alternatives classical telephone operators. VoIP calls made form VoIP device to another VoIP device are typically free but calls from VoIP to PSTN (Public Switched Telephone Network) networks are usually charged. Another cost savings are implied from the simplified network infrastructure. For example single network connectivity to carry both the voice and data, especially in situations when the internet connection is under-utilized and VoIP transmissions means no additional cost. [1]

2.2 THE DISADVANTAGES OF THE VOIP

No matter how rapidly is the Internet telephony progressing, VoIP still has problems with reliability and sound quality, due to the Internet bandwidth limitations. As a result, most corporations looking to reduce their phone bills today confine their Internet-telephony applications to their intranet. Intranet can support full-duplex real-time voice communications because it has more predictable available bandwidth than the public Internet. As the speed of Internet connections grows it will be less problem in the future.
Another problem of the VoIP is the current world of the NAT (Network Address Translation)Internet. The usage of the NAT routers brings the problems with routing VoIP traffic through this network translators.

2.2.1 BANDWIDTH REQUIREMENTS

Requirements on the internet connections are quite strong for some cases, especially when a man want to transfer more simultaneous voice calls. This problem arises with ADSL (Asymmetrical Digital Subscriber Line) when the upload transfer rate is much smaller than download. While the download speed is about 1024kbps the upload speed is usually limited to 128 kbps. This kind of the internet connection allows just one simultaneous call using the G.711 codec. At the moment, when available bandwidth is not sufficient, packet losses occurs. In voice communications, packet loss shows up in the form of gaps or periods of silence in the conversation, leading to a clipped-speech effect, that is unsatisfactory for most users and unacceptable in business communications.

2.2.2 QUALITY OF SERVICE

Because the Internet is a packet-switched or connectionless network, the individual packets of each voice signal travel over separate network paths for reassembly in the proper sequence at their ultimate destination. While this makes for a more efficient use of network resources than circuit-switched PSTN, which routes a call over single path, it also increases the changes for packet loss. The main problem is that the Internet Protocol does not provide mechanism to provide (QOS) (Quality of Service) guarantee and data packets delivering in sequential order. VoIP implementations have to solve problems with packet latencies and variation in delay called jitter. Jitter problem is usually solved by so called jitter buffer prolonging the latency on the other hand.

2.2.3 DELAYS

Delays in current telephone networks are basically divided into two kinds:

- Propagation delay caused by the characteristics of the speed of light travelling via a fiber-optic-based or copper-based medium of the underlying network.
- Handling delay caused by the devices that handle voice information and have a significant impact on voice quality in a packet network. This delay includes the time it takes to generate a voice packet. Signal processors takes from 5-20 ms to generate a frame and usually one or more frames are placed in a voice packet. Another component of this delay is the time taken to move the packet to the output queue. The actual delay at the output queue, in terms of time spent in the queue before being serviced, is yet another component of this handling delay and is normally around 10 ms. Delay inducted by the codec algorithm is considered a handling delay. It takes from 5 ms for G.711 PCM codec to 15 ms for G.729 codec. [2]
- Serializadetion delay caused by the time a router takes to place a packet on a wire for transmission. Fragmentation helps to eliminate serialization delay but it does not help with queuing mechanism in place. Conversely, if there is a queuing mechanism in place, but no fragmentation, voice traffic can still fail. Therefore, it is essential that there is a method for a router to break large data packets into smaller ones, and queuing strategy in place to help voice packets jump to the front of a queue ahead of data packets for transmission.

2.3 RELIABILITY

While the PSTN is extremely reliable the VoIP suffers for more complexity. PSTN has been matured over decades to be fail-proof, while broadband network technologies are much newer. VoIP reliability depends on the conducted reliability of the Internet connection, VoIP providers and end devices. Furthermore consumer network technologies are not subject for the SLA (Service Level Agreement) as the PSTN technologies. form of negative acknowledgement with a packet retransmission handshake for error recovery. The negative acknowledgement with subsequent retransmission handshake adds

2.4 VOIP PROTOCOLS

VoIP protocols are basically divided into signaling protocols (SIP, H.323 and others) and protocols for delivering multimedia content (as RTP). Major signaling protocols used in VoIP are SIP (Session Initiation Protocol) and H.323. H.323 is traditionally more associated with the ITU (International Telecommunication Union) while SIP has it's roots in the internet community

3 CONVERGENCE.

3.1 REAL-TIME TRANSPORT PROTOCOL (RTP)

The RTP defines a standardized packet format for delivering audio and video over the Internet. RTP does not have a standard TCP or UDP port that it communicates on. Although there are no standards assigned, RTP is generally configured to use ports 16384-32767. RTP only carries voice/video data. Call setup and tear-down is usually performed by the SIP or other protocol. The fact that RTP uses a dynamic port range makes it difficult for it to traverse firewalls. In order to get around this problem, it is often necessary to set up a STUN (Simple Traversal of UDP over Nets) server. It was originally designed as a multicast protocol, but has since been applied in many uncast applications. It is frequently used in streaming media systems as well as video conferencing and push to talk systems, making it the technical foundation of the Voice over IP industry. It goes along with the RTCP and it's built on top of the User Datagram Protocol (UDP). Applications using RTP are less sensitive to packet loss, but typically very sensitive to delays, so UDP is a better choice than TCP for such applications. The protocols themselves do not provide mechanisms to ensure timely delivery. They also do not give any Quality of Service (QoS) guarantees. These things have to be provided by some other mechanism. Also, out of order delivery is still possible, and flow and congestion control are not directly supported.

3.2 H.323

The H.323 standard has been developed by the ITU-T for equipment manufacturers and vendors who provide Voice over IP service. It provides technical recommendations for voice communication over LANs assuming that no Quality of Service (QoS) is being provided by LANs. It was originally developed for multimedia conferencing on LANs, but was later extended to Voice over IP.

3.3 COMPONENTS OF H.323

The H.323 standard proposes an architecture that is composed of four logical components – Terminal, Gateways, Gatekeepers and Multipoint Control Units (MCUs). The architecture schematic is depicted in the following diagram. The various components are described subsequently.

3.4 TERMINALS

These are LAN client endpoints that provide real-time, two-way communications. All H.323 terminals are required to support H.245, H.225, Q.931, Registration Admission Status (RAS) and real-time transport (RTP) protocols. H.245 is used for controlling channel usage, while H.225 or Q.931 are used for call signaling, call setup and tear down. RTP is used as a media transport protocol that carries the voice traffic. RAS is used by the endpoint for interacting with the gatekeeper. H.323 terminals may also use T.120 data conferencing protocols, video codes and support for MCU. An H.323 terminal can communicate with either another H.323 terminal, a H.323 gateway or a MCU.

CONCLUSION

As the Internet has revolutionized the nature of business and social live, the Internet penetration grows exponentially. As the available internet bandwidth grows it is clear, that VoIP technology is pushing out standard circuit switched technology and in the near future will be switched networks completely obsolete. However, VoIP still suffers from packet transmission related problems like jitter, network congestion and so on.

My project was concerning with general analysis of VoIP technology and has mentioned some related network problems and their solutions. In practical part I have configured Linux private branch exchange based on the Asterisk project. It consisted of one PSTN ISDN line, one old ISDN private branch exchange and one VoIP phone interconnected one another with an Asterisk software to be an integrated private branch exchange. PSTN ISDN line was connected over ISDN interface card Fritz!Card DSL running in TE mode and old Fritz!X PBX was interfaced with a HFC-S based interface card running in NT mode. I have also configured traffic controlling script for prioritization VoIP traffic over normal non real-time traffic. Asterisk exchange is very complex and complicated systems and so are complicated all the protocols.

REFERENCES

[1] Voice over IP.
URL: <http://en.wikipedia.org/wiki/Voip> [cited 2006-06-25]

[2] MITRA, D. Network Convergence and Voice over IP. Sunnyvale: Tata, 2001.

[3] MADSEN, L. et al. Asterisk: The Future of Telephony. 1st edition. Sebastopol: O'Reilly, 2005. 404 pages. ISBN: 0-596-00962-3

[4] *Skype.*
URL: <http://en.wikipedia.org/wiki/Skype> [cited 2006-06-24]