Advantages of VoIP in the german research network

Falko Dressler

Regional Computing Center (RRZE) / Department of Computer Science IV (Operating Systems)
University of Erlangen-Nuremberg
Martensstrasse 1, 91058 Erlangen, Germany
Phone: + 49 9131 85-27802, Fax: +49 9131 302941
E-mail: falko.dressler@rrze.uni-erlangen.de / fd@acm.org

Abstract

With the arising of telephony over the internet, loads of institutes start looking at that new technology. They ask themselves for sense and meaningful use. Many questions appeared to be answered first. The first one is about the interoperability of the end systems. It should be possible to change the vendor and implement the new devices without any problems into the existing VoIP infrastructure. These ideas led to first single VoIP labs. To increase the value of such installations, they should be interconnected. Because of the very heterogeneous hardware, which has been used by the institutes, the construction of a VoIP network has been a project for itself. Several working groups have been formed in order to examine single questions much closer. Examples are the accounting, the security and the required quality of service within the transport network. However, the research work has not finished yet. There are still unsolved questions about the accounting and the security in such a large VoIP network as it became today. Routing strategies have to be discussed as well as security questions. Maybe a single institution should start offering IP telephony as a service for the complete german research network. Currently, it is a distributed group.
During the tests, VoIP has proved its position within the internet landscape as well as within the standard telephony. The participants of the VoIP project managed to configure and use the VoIP components successfully. Nevertheless, it should not be concealed that much work is still approaching.

1 Introduction

In 2000, three german universities started to build small labs to test Voice over IP applications. Each one installed its own gateways, hard-phones as well as software based telephony applications, mainly based on H.323 for signaling.

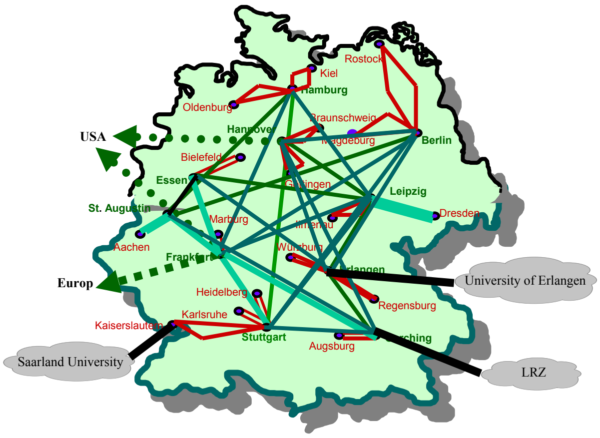


Figure 1 G-WiN including initial VoIP participants

All these labs have had different goals. One team started measurements of the QoS (quality of service) requirements of VoIP but another one specialized on the management and the accounting.
Soon, all groups realized the benefit of interconnecting these labs. This was the beginning of the VoIP network in the german research network (G-WiN). The first members (figure 1) were the Saarland University, the University of Erlangen-Nuremberg and the LRZ (Leibniz Rechenzentrum, interconnects the universities in and around Munich).
Research results have published at various conferences and in different presentations [2], [3] and [4]. That aroused the interest of other universities, companies and manufacturers of VoIP equipment. The result was a much broader VoIP project with a number of participating universities as well as the leading manufactures of VoIP hardware and software. Meanwhile, about 10 universities interconnected their VoIP networks. The 'lab' grew to a useful network with very heterogeneous products. Most members also connected the IP telephony clouds to their 'normal' telephony system. This allows the use of the G-WiN for data transfers as well as for telephone calls.

The following chapters should summarize the VoIP project. In the first part, information is provided about the configuration of the VoIP network and the local installations as well as about the management of this network. The next chapter informs about the executed QoS measurements. A final chapter provides an overview about future activities.

2 Configuration

The following figure (figure 2) should give an overview over the typical local VoIP infrastructure.

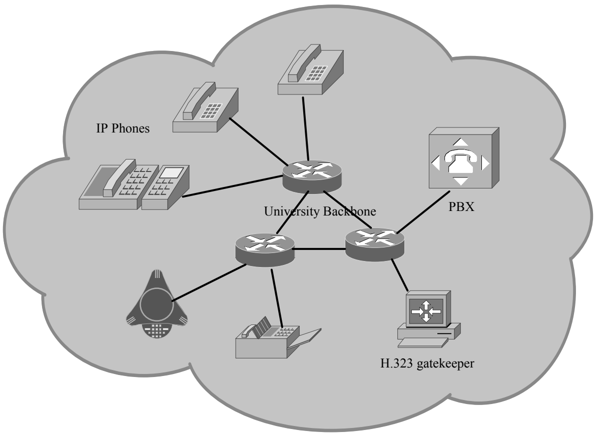


Figure 2 Local VoIP infrastructure

Most participants of the VoIP network are using the Cisco Callmanager as the gatekeeper as well as to manage their Cisco IP phones, but systems from Alcatel, Siemens and Wiptel are also part of the test equipment. For the clients, mostly H.323 devices are in use. Examples are various hard-phones from Cisco or Siemens as well as soft-phones like Windows Netmeeting. Different types of gateways build the interconnection to the POTS (plain old telephony system, the standard university telephone system).

It is to say, that some of the installations are used for both, tests within the VoIP network as well as for real application.

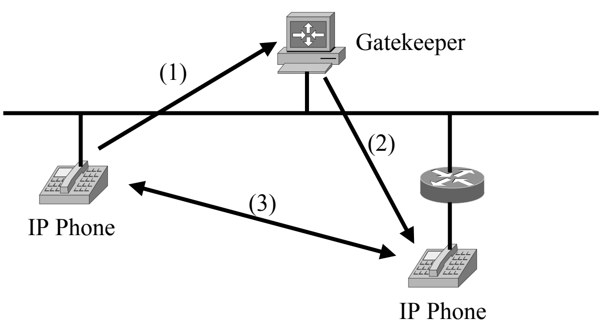
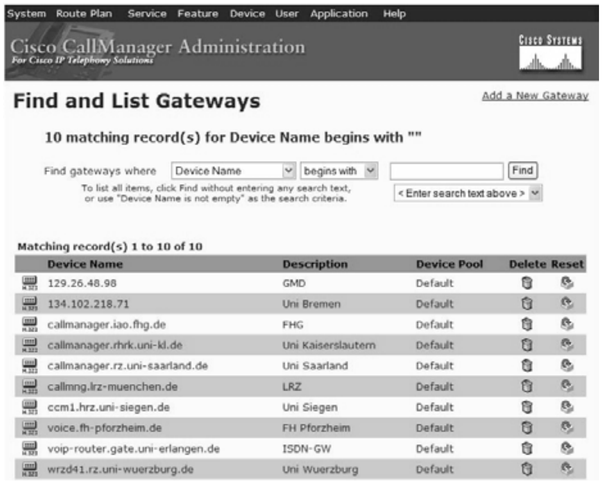


Figure 3 H.323 communication

Figure 3 shows the basic operations to set-up a call using H.323. First (1), the calling end system contacts the gatekeeper to establish the call. In a second step (2), the gatekeeper connects to the destination phone. This can also be a gateway in a longer row of interconnected VoIP clouds. After this initialization (3), the VoIP devices intercommunicate directly to each other without interaction and regardless of the proper function of the gatekeeper.

Gateways have to be deployed to interconnect single H.323 installations. Figure 4 shows a screenshot of the Callmanager configuration showing the currently installed gateways. The common policy within the VoIP group is to allow connections from and to each participant.



Device Name	Description	Device Pool	Delete	Reset
129.26.48.98	GMD	Default		
134.102.218.71	Uni Bremen	Default		
callmanager.iao.fhg.de	FHG	Default		
callmanager.rhrk.uni-kl.de	Uni Kaiserslautern	Default		
callmanager.rz.uni-saarland.de	Uni Saarland	Default		
callmng.lrz-muenchen.de	LRZ	Default		
ccm1.hrz.uni-siegen.de	Uni Siegen	Default		
voice.fh-pforzheim.de	FH Pforzheim	Default		
voip-router.gate.uni-erlangen.de	ISDN-GW	Default		
wrzd41.rz.uni-wuerzburg.de	Uni Wuerzburg	Default		

Figure 4 H.323 gateways in the VoIP network

Questions about the numbering scheme and the VoIP routing with or without redundancy are discussed in chapter 3.

The main problem in interconnecting all the telephony systems is the very heterogeneous hardware and software. The most frequently discovered problems are the interoperation of VoIP equipment of different manufacturers. Then, all the gatekeepers have to internetwork together. Based on the H.323 standard both problems should be solvable. Unfortunately, not all the H.323 devices are interoperable due to missing or incorrect implemented functions of the H.323 standard. Additionally, each manufacturer has implemented proprietary features, which of course do not work between different implementations.

The most difficult function is to provide the connectivity to the POTS of the institute. Since most universities use very old installations of telephony systems, the interconnection to the VoIP network is generally unsolved.

3 Management

During the first tests, it showed up that there are a large number of management functions. Some of these are only of local importance, so each institute can apply its own conceptions and restrictions. Examples of local configurations are local directory services, IP parameters and the communication between the clients and the gatekeeper.

Other management questions have an impact on the internetworking between the local installations. There has to be a consensus on these items. First, all the participating groups have to apply to a unique numbering system. Initiated by the University of Erlangen-Nuremberg and the Saarland University, these terms have been centralized. Supported by the TZI (Center for Computing Technologies) at the University of Bremen, a web page was installed, which describes the numbering system of each participant, the used gatekeeper and how to interconnect via H.323. Figure 5 shows an excerpt of the numbering system.

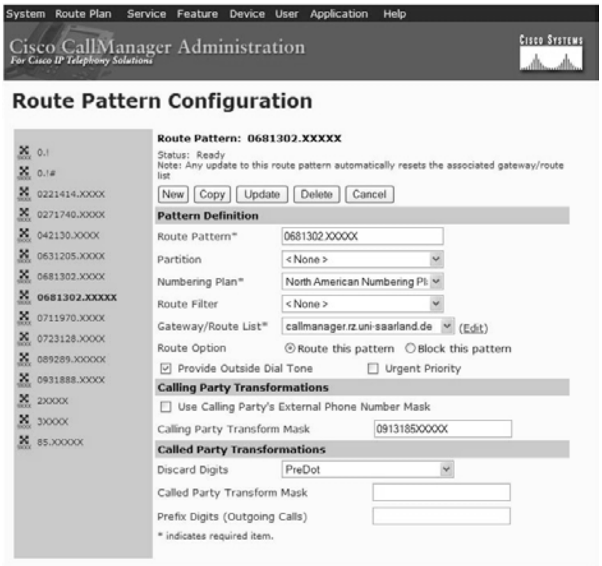


Figure 5 Route pattern in the VoIP network

Currently unsolved is the question how to implement a useful accounting, which operates between all the participating institutes. Involved is also a political problem. A first idea could be to use the VoIP network to save costs for long distance calls. Due to the non-uniform accounting strategies of the different vendors as well as the non-uniform ideas of the participants how to account, the use of the VoIP network for calls outside the institutes is very restricted. A working group led by the Saarland University is researching on these questions.

A goal of some of the tests within the VoIP network is to get information about the principles and best current practices of the VoIP routing. Figure 6 shows four VoIP clouds which should be hooked together. Network 1, 2 and 3 are full meshed. This allows tests of alternate routing strategies. Network 4 can only be reached via network 3. Therefore, network 3 is a transit-network within the VoIP network.

Tests have shown that there are no problems to make use of transit-networks except of security problems that are discussed later in chapter 5.

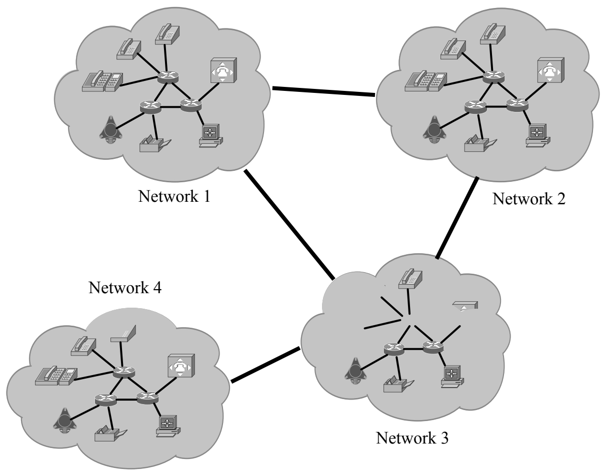


Figure 6 Types of cross-linking

The VoIP installation of the University of Erlangen-Nuremberg has been used for a long time as a cross-connector between all the other VoIP clouds without any problems. Of course, there was no redundancy for the single gateway in Erlangen, which interconnected each participant. Right now, the VoIP group has configured the network full-meshed. This prevents problems that occur due to failures of single gateways.

4 QoS measurements

One very first question was 'How much QoS is required for VoIP'. At the University of Erlangen, a research group started with different measurements on this topic. Lab tests are involved as well as tests over the campus network and the G-WiN.

Based on the working principles of the IP based internet, the tests include measurements of values such as the maximum absolute delay, the maximum delay variation (the maximum jitter) and the maximum packet loss ratio. Together, all these values describe the minimum QoS requirements of VoIP on the underlying IP network.

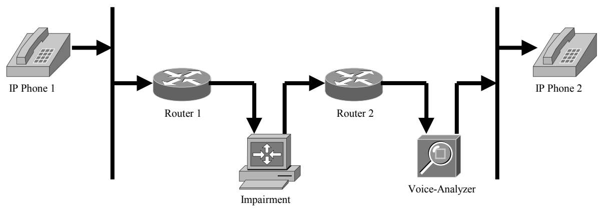


Figure 7 Test environment for QoS measurements

Figure 7 shows the principles of the lab measurements. A laboratory environment has connected two IP phones. This environment contains an impairment tool, which was used to simulate different behaviors of real networks such as a constant or variable packet loss or a constant or variable delay. The impairment tool (GN Nettest Interwatch 95000) has been designed for measurements in ATM networks. Two Cisco routers are used to convert the ethernet frames to ATM cells and vice versa for the impairment tool to work. A voice analyzer (RADCOM) is used to get objective results from the tests.

During the tests, one very important point was to correlate objective measurement results and subjective impressions. Mostly, both test methods have shown the same results. Results from the measurements are shown in table 1.

Table 1 VoIP and packet loss

cell loss ratio [%]	packet loss ratio [%]	subjective impressions
0	0	good quality
10-4	0,069	good quality
10-3	0,785	some interferences
10-2	8	unusable for music, stong interferences for voice
10-1	72,3	unusable for voice transmissions

Measurements of a constant delay have shown no lost packets and always a very good quality. Only the bidirectional communication is slowing down or getting unusable due to high delays in discussions. The introduction of a high jitter (delay variation) results in lost packets because of the working principles of RTP (Real-time Transport Protocol, [6]). This is because packets, which are too late, are just dropped rather than queued. For real-time multimedia transmissions it is much more important to get the packets (sequences, frames, etc.) as fast as possible, than to get every packet. Queuing up the packets would result in a much larger overall delay but dropping single packets just introduces small disturbances, which are almost tolerable. Therefore, the results in table 1 apply to jitter measurements as well.

In addition, stress tests have been done by overloading the CPUs of the routers or by oversubscribing the available bandwidth at the outgoing interface of the router. Most results have been already published [3], [4] and [5].

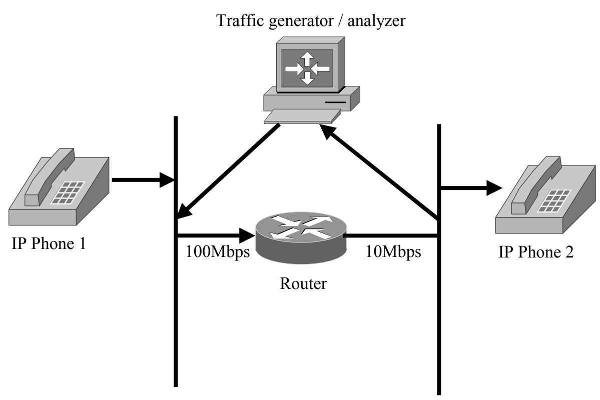


Figure 8 Test environment for stress tests

The test environment is shown in figure 8. A single router connects the two IP phones. A traffic generator (HP4200B) has been used to simulate background traffic, which is freely configurable. The maximum data rate at the outgoing interface of the router has been fixed to 10Mbps. This allowed to overload this interface and forced the router to drop packets. The stress tests included the activation of QoS mechanisms at the routers. First tests have been done using a Cisco router but the research group decided to switch to a Linux system because the required queuing strategies are implemented on this system [1]. In addition, the source code of the system is freely available in order to examine exact functionality.

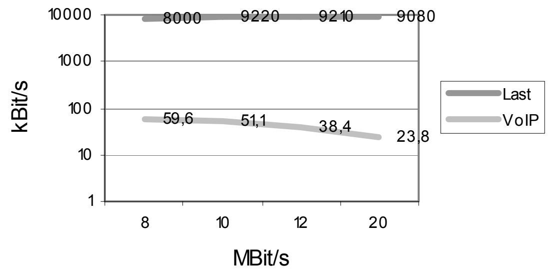


Figure 9 FIFO queuing

Tests have been done using different queuing strategies. For a first and reference measure, the router has been configured to use its standard FIFO (first-in, first-out) queuing mechanism. The results of this test are shown in figure 9. With increasing background traffic, starting at 8Mbps, continuing with 10 and 12Mbps and ending with 20Mbps, the voice traffic was reduced down to 23.8kbps out of 59.6kbps. Already with background traffic of 12Mbps, the telephony application was unusable due to the maximum delivered rate of 38,4kbps.

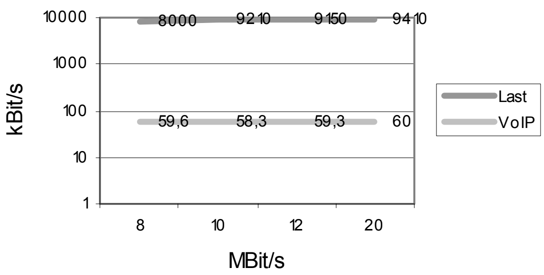


Figure 10 CBQ

The next tests involved a queuing strategy, which is recommended for real-time traffic, CBQ – class based queuing. The traffic has been classified on basis of the IP source address. Mechanisms to prevent overload on the incoming interface such as RED (random early detect) or WRED (weighted RED) have not been used because all the data streams used UDP (User Datagram

Protocol) which is not affected by them. Two queues have been configured to take up packets of the different traffic streams. Each queue has been assigned the same priority. So no priority queuing was involved, which could improve the results still a little more. Figure 10 shows the measured values. It can be seen that the increasing background traffic does not affect the voice traffic.

5 Future activities

Currently, the members in the VoIP project think about an enlargement of their VoIP network. First, the local installations at each participating university grow and second, other universities, research institutes but also some companies in Germany want to join the project.
There are various problems, which need to be solved by several working groups in the project. The most important one is to implement an accounting accepted by each member. The problem thereby is the very heterogeneous installed VoIP hardware. In addition, there are more political questions to be solved between the participants. These are questions such as 'Can we allow other institutes to use our telephony infrastructure? And if we do it, for which charge? '.
Based on the QoS measurements, new network technologies have to be implemented into the local campus networks as well as into the network infrastructure of the german research network. One example is the differentiated services architecture including classification schemes, mechanisms to prevent overloading the input interfaces and queuing strategies for real-time services. This is a straightforward approach in each local network but fails currently in carrier networks such as the german research network.
Another working group is working on security questions, which appear to become more and more important while the VoIP network is growing. A first question is whether one can and who is allowed to use which resources. Especially transit-networks (in terms of voice over IP) have to be protected. Last, but not least, questions about encryption mechanisms for the voice transmission have to be solved. Currently, there are available algorithms and methods developed by the IETF (Internet Engineering Task Force) but no vendor has implemented any f them yet.

6 Summary

Summarizing the VoIP project it can be said, it works. Additionally, it works for small as well as for larger installations. The participants use the VoIP network besides the measurements, tests and developments successfully for standard telephony.
It has proven that most devices interoperate successfully based on standards as H.323. Some vendors have implemented proprietary features like voice-mailboxes, the display of the name of the caller or corporate directories, which can be used only within the common equipped VoIP cloud.
The VoIP project includes working on very different items by various research groups. This allows sharing a highly specialized knowledge between all the participants in this project. In addition, there are knowledge centers at the universities and institutes, which get involved into the research process. This allows a synergetic use of existing resources.

References

[1] W. Almesberger, "Linux Network Traffic Control – Implementation Overview," Website: http://icawww1.epfl.ch/linux-diffserv/, 1999.

[2] F. Dressler, U. Hilgers, S. Nägele-Jackson, K. Liebl, "Untersuchung von Dienstqualitäten bei echtzeitorientierten multimedialen Datenübertragungen," In: Multimedia und Automatisierung: Fachtagung der GI-Fachgruppe 4.4.2 Echtzeitprogrammierung, PEARL, Boppard, 25./26. November 1999 / PEARL '99, Workshop über Realzeitsysteme. Peter Holleczek (Hrsg.). Springer, Berlin, 1999 (Informatik aktuell).

[3] F. Dressler, U. Hilgers, "Routing mit QoS-Eigenschaften unter Linux," In: Echtzeitbetriebssysteme und LINUX: Fachtagung der GI-Fachgruppe 4.4.2 Echtzeitprogrammierung, PEARL, Boppard, 23./24. November 2000 / PEARL 2000, Workshop über Realzeitsysteme. Peter Holleczek (Hrsg.). Springer, Berlin, 2000 (Informatik aktuell).

[4] F. Dressler, U. Hilgers, P. Holleczek, "Voice over IP in Weitverkehrsnetzen?," In: Anwendungs- und System-Management im Zeichen von Multimedia und E-Business: Fachtagung der GI-Fachgruppe 3.4 Betrieb von Informations- und Kommunikationssystemen, BIK, Tübingen, 4.-6. April 2001 / BIK 2001.

[5] U. Hilgers, F. Dressler, "Echtzeit-Datenverkehr über IP-basierte Datennetze," In: Echtzeitkommunikation und Ethernet/Internet: Fachtagung der GI-Fachgruppe 4.4.2 Echtzeitprogrammierung, PEARL, Boppard, 22./23. November 2001 / PEARL 2001, Workshop über Realzeitsysteme. P. Holleczek, B. Vogel-Heuser (Hrsg.). Springer, Berlin, 2001 (Informatik aktuell).

[6] H. Schulzrinne, S. Casner, R. Frederick, V. Jackobson, "RTP: A Transport Protocol for Real-Time Applications," RFC1889, IETF, 1996.