Voice over IP in the German Research Network: Challenges and Solutions

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***Abstract* - The goal of this paper is to provide some feedback on a VoIP project in the German research network. During the last couple of years we build up and managed an unusual large testbed for VoIP applications in Germany. Several universities, research institutes, and companies tied together to allow a common usage to different local VoIP installations. The primary goal of the tests was to identify problems emerging in such large installations. We evaluated the quality of service requirements of VoIP transmissions as well as to examine the interoperability of different VoIP components. Additionally, as a very main part of the investigations, the call routing possibilities were tested including redundancy options. This resulted in a reconfiguration of the whole VoIP network several times. A short discussion on current security questions around the voice over IP employment is rounding up the paper.**

***Keywords* - Voice over IP (VoIP), Service Quality, Quality of Service Requirements, H.323**

1. Introduction

During the last three years, several universities and institutes in Germany set up a test bed for several IP telephony applications. The project started in 2000 with different implementations at several research institutes including the University of Erlangen-Nuremberg and the Saarland University. During the first months, a common playground has been formed by interconnecting all the single labs. Using the new Germany-wide Voice over IP (VoIP) network, we were able to initiate a couple of sub-projects. The focus of these projects was to figure out the quality of service requirements of VoIP, the interoperability of the available installations, the optimum routing properties, and the inclusion of gateways between different protocol versions.

The goal of this paper is to provide an overview to the activities which have been taken since the test bed was formed. Additionally, the identified challenges and solutions are discussed. Today, the test bed is moving from a lab environment to an operational system which interconnects the telephone systems of various German universities, research institutes, and companies. Due to this change, it is losing its research character even if it is still possible to use the network for tests of new components and technologies.

The following issues are presented and discussed within this paper:

- Motivation for the project and current status
- Quality of service (QoS) requirements
- Interoperability questions
- Telephony routing and redundancy
- Security considerations

The current status of the project is a smooth movement from a research network into an operational state. Therefore, the University of Erlangen-Nuremberg decided to declare the academic phase as finished.

2. The Project: Motivation and Status

The project was initiated in order to increase the possibilities to analyze voice over IP applications [6]. The first participants were the Saarland University and the University of Erlangen-Nuremberg. The extension of the single labs by building a Germany-wide VoIP testbed led to a number of new options.

The basic motivation was to test the capabilities of voice over IP in general and the quality of service requirements in particular. Additionally, it was the goal to identify the challenges in building large VoIP installations and interconnecting them.

All the participants of this project are connected to the German research network (G-WiN) which is the counterpart of the Abilene network in the United States. All the German universities are interconnected by the G-WiN with bandwidths up to 2.5 Gbps. Therefore, we have had a quite impressive testbed with no real bottle necks available for the tests.

At the end of 2002 the focus of our initiative changed from a research point of view to an operating telephony network with higher availability demands. Certainly, the network cannot be used as a testbed only anymore. Therefore, we considered it the best time to summarize the activities and results of all the single projects using the VoIP testbed.

3. Quality of Service Requirements

The main focus of the research work in Erlangen was on the quality of service requirements of voice transmissions over the internet. The typical VoIP telephones employ H.323 as the signaling protocol [1]. This document of the ITU contains a framework for different standards to transport multimedia content over IP networks. With this, a protocol is presented to define the functionality for equipment of different vendors. The multimedia data are transported employing the real-time transport protocol (RTP) over the IP network [10]. This protocol provides end-to-end network transport functions to support real-time multimedia applications. To monitor the delivery of data RTP is accompanied by the real-time control protocol (RTCP). RTCP contains a QoS feedback mechanism which informs the sender about the current transmission quality at the receiving side, e.g. the jitter or the number of lost packets.

In our lab we installed a small VoIP cloud consisting of several telephones and the Callmanager from Cisco, Inc. that controls the telephone functions of the Cisco phones as well as to provide routing facilities for standard H.323 clients.

The test setup for the QoS measurements is shown in Fig 1. There were two ethernet LANs to which the VoIP telephones are hooked up to. Both LANs are connected by two Cisco routers. The network between them is a STM1-ATM link, which represents a WAN connection. In this connection an INTERWATH 95000 was installed as an impairment tool to reproduce the typical behavior of wide area networks, for example by introducing an artificial delay or a configurable packet loss ratio. The measurements have been done using a RADCOM voice analyzer. It is able to distinguish between text-based data and speech. It computes information about jitter, packet loss, silence suppression, and packets which arrive out of order.

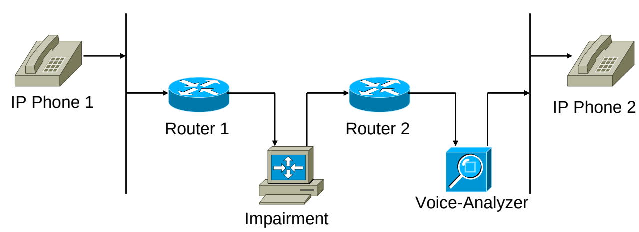

Fig 1. Test setup used for all the lab measurements. The impairment tool was used to modify the networks behavior in a controlled way. The voice-analyzer was included to achieve objectively comparable results.

In all the tests we transmitted both, classic music and speech over the telephone line. With this, we were able to find out which objectively measured data are important for which kind of voice transmission – bidirectional communication and unidirectional broadcast – and what are the main differences between music and speech. The taken results of the tests were already published at various conferences [5, 6, 9] and should be summarized at this place.

Table 1 Behavior of VoIP on 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, strong interferences for voice
10^{-1}	72.3	unusable for voice transmissions

The tests showed the QoS needs of VoIP. Increasing delay has only a great impact if the voice transmission is bidirectional (data not shown). But for unidirectional broadcasts the listener can only recognize little impairment. A more serious problem is caused by delay variation, which can lead to lost packets. Little losses or bit errors cause great impairment up to unintelligibility in voice transmission. In Table 1 a comparison between objective measurements (packet loss ratio) and subjective impressions (as observed by some end user) is provided. Together, all these tests showed that just a small degradation of the available quality of service results in an unintelligible call.

In another test we checked the influence of a congestion situation on the quality of the voice traffic. We used the same test setup as before, but an additional traffic source sent data from router 1 to router 2 as shown in Fig 2. This traffic is routed over the same line as the voice traffic, thus influencing the voice transmission. The additional background load was able to produce an overload situation at the outgoing interface of router 1.

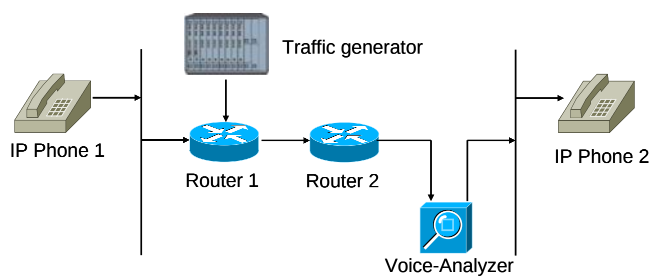

Fig 2. Test setup to check the behavior of telephony in congested networks. The traffic generator sent a background load into the lab network.

In the first test, the background load was increased continuously. At the situation, when packet losses occurred at the outgoing interface of router 1, the quality of the voice quality degrades. With this one can see, that the congested IP network with standard algorithms has not been able to fulfill the quality needs of voice data (see Fig 3). In summary one can say that the IP protocol as it is seems not to be a good approach to transport real time multimedia data. Concepts are needed which add QoS functionality to the IP protocol. Therefore, we finally used the same test environment to examine the currently available mechanisms in IP networks to enhance the service quality or even to guarantee some minima.

One approach to do this is the Differentiated Services concept [3]. It adds a scalable mechanism to the IP protocol in order to provide different service classes to different applications. To assign a packet to a service class, it is marked with a corresponding tag in the IP header. According to this header value, traffic conditioning functions like classifying and policing are performed. In addition, this value determines the forwarding per hop behavior (PHB) in the network nodes corresponding to the negotiated QoS for the special service class. The PHB is implemented by resource and queue management mechanisms, including scheduling and congestion avoidance algorithms. Scheduling algorithms can be used to reduce the jitter in the outgoing queues on the router interfaces or to give special IP packets priorities or a special amount of buffer space. One suitable algorithm is Class Based Queuing (CBQ). CBQ is a link-sharing mechanism which allows multiple traffic classes to share the bandwidth on a link in a controlled fashion [7] in contrast to the standard scheduling mechanism FIFO (First in, First out). There the packets are served in the order, in which they arrive.

For the tests we focused mainly on the examination of the capabilities of the differentiated services model [3] implemented in various routing systems such as the Cisco IOS and the Linux operating system [2]. We repeated the test described above. The background load was increased continuously and at the same time, CBQ was activated on the outgoing interface of router 1. With this it can obviously be seen, that the background traffic has no longer any impact on the voice traffic. The throughput of the voice traffic remained constant, i.e. no packet losses occurred, and the delay did not increase. Also the subjective impressions of these tests are comparable with the objective results. Detailed measurement results can be found at [5, 6]. Therefore, congested IP networks with the standard IP protocol with FIFO queuing seem not to be a good approach to transport real time multimedia data. They are not able to fulfill the quality needs of voice data. It was shown that queuing mechanisms like CBQ help to increase the quality of service in the network in such a fashion that IP telephony applications can be used without any quality detraction [4, 5].

4. Interoperability Questions

Most of the leading companies in the area of network and telephone systems recognized the growing market for VoIP components and solutions. Therefore, various systems have become available. One part of our tests, shown in Fig 3), was to interconnect the different sites running different installations.

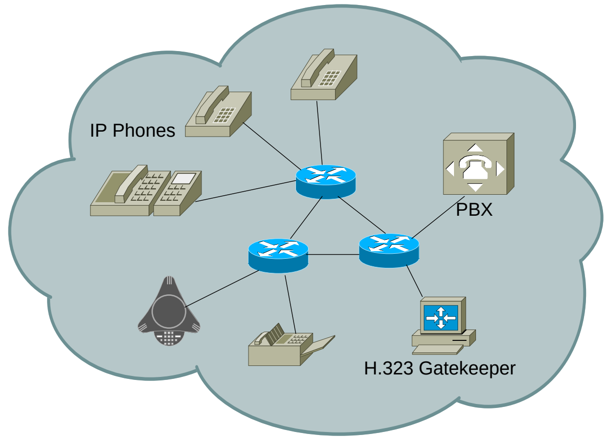

Fig 3. Interoperability Tests. First, shown on the left side, between a VoIP network and a POTS system which is attached via a PBX (private branch exchange), secondly, shown on the right side, between different VoIP networks.

For example, the most well-known systems from Cisco and Siemens have been installed at multiple locations in the testbed. Based on the standard signaling protocol H.323 [1] an interoperating of all the devices has been achieved. Nevertheless, various problems have been identified. Using H.323, only a very basic internetworking can be offered. We included devices from different companies in our interoperability tests. Examples are hardphones from Cisco and Siemens, software based telephony applications from Cisco and Microsoft.

It has been shown that the different phone can be interconnected very easily using the standard H.323 signaling protocol. The process of integrating new devices was mostly straightforward. Typically it just required the configuration of the IP address and the local numbering scheme to interconnect a new client. Also, there was no difference between using software based applications such as the Microsoft Netmeeting application and employing hardphones such as ones from Siemens.

Using proprietary mechanisms of the devices, the functionality of the IP phones can be dramatically increased. Standard telephone applications using the POTS (plain old telephony system) environments are offering such functionality and, therefore, the end users demand the same behavior. The second major signaling protocol for voice over IP is SIP (session initiation protocol, [8]). It has some advantages which were examined in some tests.

Unfortunately, H.323 still is the standard protocol and both are incompatible. Therefore, gateways are required between each protocol worlds. There are numerous groups working on this topic.

Secondly, the IP telephony systems must be connected to the standard telephone systems at each participant for several reasons. First, the interconnectivity with the local telephone system should be tested and, secondly, it is a strong requirement to be able to place calls from inside the VoIP domain to phones outside the IP world and vice versa. Such a scenario is shown in Fig 4.

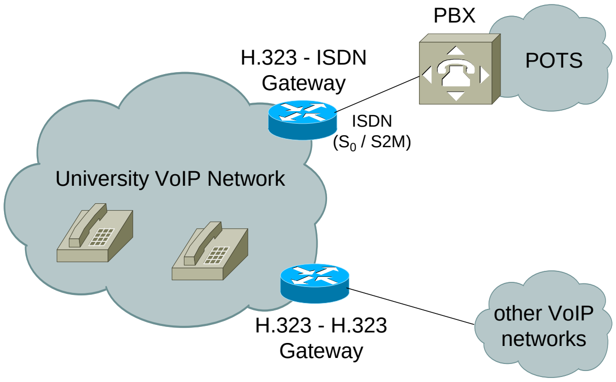

Fig 4. Typical setup to test the interoperability between the VoIP network and the classic POTS and towards other VoIP installations.

For this purpose we employed a Cisco 2610 which was router connected to the public telephone network using an ISDN S_0 connection which provides two separate channels for ongoing calls. Other institutes used to employ similar routers typically using S2M interfaces to their local university telephone networks. It occurred that the most difficult step is the interconnectivity of the VoIP network with the local PBX at the institute. Sometimes there were technical problems to solve – which the manufacturing companies were doing a tough job to finish in time – but mostly there are political questions blocking the way due to different responsible organizations for the network facilities and the telephone infrastructure.

5. Telephony Call Routing and Redundancy

Other terms of research in the project were the call routing facilities. In a very local VoIP installation this is a straightforward configuration step. Typically, all the local phones are hold in a database on the gatekeeper. Long distance calls, or more precisely all the calls to non-IP phones, are routed to a gateway which points to the POTS.

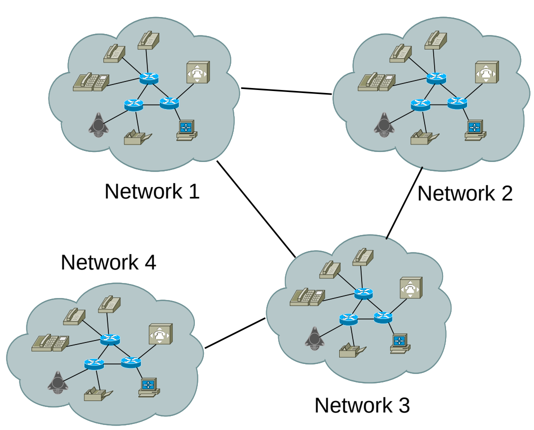

Fig 5. Setup possibilities for interconnections of VoIP clouds.

The more flexibility is included into the VoIP installation the more complex is the resulting routing configuration. This applies even stronger to our VoIP network which interconnects multiple universities and research institutes all over Germany (see Fig 5).

We figured out different possibilities to implement such a distributed system. First, we started with a central approach. All the sites installed a default route for their calls to go to a gateway in Erlangen. There we implemented a large routing table, which of course is always in a consistent state just because only one place exists where all modifications are applied. Secondly, to enable a much higher redundancy – for the routing core as well as for the availability of the network connections – the routing decisions were moved to the local gatekeeper of each institute. To get this principle work a full meshed VoIP network is required and every participant has to configure a path for each other. Rapidly it has been shown that the complexity of the management of this approach is much too high. Therefore, a third and maybe even best approach was started. It concept is to have about three central routing cores containing the complete routing information and to apply three default routes at each single VoIP network to these core gateways. This allows a high availability compared with a low complexity.

Nevertheless, at the moment a central gatekeeper is installed to obtain the routing table. All the calls are routed via this system. In case of a failure – if the gatekeeper is down of just not reachable at all – the calls are rerouted using the public telephone network. This might not be the optimum solution but it has proven to be a working one which does not rely that much on a real cooperation between all the participants and, therefore, prevents any political treats.

6. Security Considerations

The last section is intended to provide a short overview to the current security issues in VoIP networks. These issues may be divided into three groups:

- Security of a single connection, i.e. the possibility to encrypt the session to prevent eavesdropper to watch the call. As VoIP traffic is transported over a data network where many users have access to, theoretically every one of them can try to identify the voice packets. As a great number of attacks occur from the inside, this can become really a problem.
- Accessibility of the VoIP network, i.e. the resources should be secured from unauthorized access. Besides access, which should be controlled, accounting is playing a large role in this subject. Even if calls through the internet may introduce low or even no costs, calls running through gateways, i.e. to the POTS, typically do have to be paid for.
- Privacy protection, i.e. the prevention of personal data being transmitted over the network. This also is a critical point for accounting reasons.

In discussions with the manufacturers of the current VoIP components, we discovered that only a few of them have realized security as an important issue. Most move the point to the network itself saying that other mechanisms such as VPNs (virtual private networks) can do the job. This is true if the VoIP network is meant to be only a local one, i.e. it has no (IP) connections to other sites. In the case of our VoIP compound in Germany, we cannot rely on such mechanisms. Unencrypted telephony calls are very easy to eavesdrop, to record, and to play back. The same applies to the access control mechanisms. But it should be mentioned here, that all available signaling protocols, which enable the user to place voice over IP calls, contain security mechanisms. For example, H.323 is accompanied by the protocol H.235 for these steps.

Finally, the potential vulnerability of the VoIP systems has to be mentioned. As e.g. the VoIP phones or the callmanager contain remote-accessible code which can be exploited. Some of the phones, e.g. Cisco 7900, include a built-in web server. This server provides pages with critical content without any authentication, e.g. debug and status information about the phone which can be used for management purposes are shown there.

7. Summary

The paper was giving a short overview to the activities around voice over IP in the German research network from the Erlangens point of view. It has been proven that VoIP is already an enormous tool allowing to build up telephony networks using the same network infrastructure which is available at all universities and research institutes for data transmissions. Several problems have been discovered and discussed such as the call routing problems and the interoperability questions. Especially in Erlangen, the requirements on the transmission quality were examined and it was shown that – even if some modifications on the current IP infrastructure are required, e.g. the implementation of priorities for different kind of network traffic – the telephony applications can well exist besides the data transmission without strongly effecting each other. Reliability is an important issue, therefore, redundancy solutions were examined, especially in the case of routing redundancy. Still unsolved is the requirement for more security. Even it might be acceptable to have unencrypted VoIP transmissions, the demand for secured accounting processes is very high. It seems that this issue is the most important one in the next future. To conclude our work it must be said that we managed it to build up a really large testbed for VoIP applications which directly got into business use. It will show up if it will survive or even strike over the good old telephony system.

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