Set of IP Telephony Best Practices in National Research Networks in EU

Best Practice Document

Produced by CESNET led working group on Multimedia transmissions and collaborative environment
(CBPD146)

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March 2011

Document No:	GN3-NA3-T4-CBPD146
Version / date:	March 7, 2011
Original language :	EN
Original title:	"Set of IP Telephony Best Practices in National Research Networks in EU"
Original version / date:	V1.2 of March 7, 2011
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The research leading to these results has received funding from the European Community's Seventh Framework Programme (FP7/2007-2013) under grant agreement n° 23 8875, relating to the project 'Multi-Gigabit European Research and Education Network and Associated Services (GN3)'.

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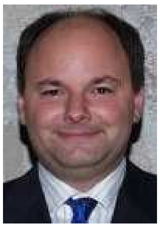

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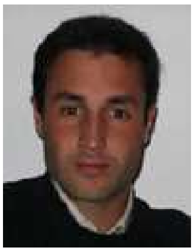

Miguel DUARTE graduated in Electronics and Computers Engineering, telecommunications branch in FCTUC – Faculdade de Ciências e Tecnologia da Universidade de Coimbra, in 2004. During his graduation he spent one year studying under the Socrates/Erasmus program in Delft University of Technology – The Netherlands (2002 – 2003). Miguel started working at Siemens Networks S.A./Nokia Siemens Networks Portugal S.A. in April 2005 as System Integration engineer where his main responsibilities were focused on IMS (IP Multimedia Subsystem) and FMC (Fixed Mobile Convergence) integration, support and technical management. In February 2008 he joined Novabase Consulting as Senior Consultant/Technical Support Manager where he was responsible for the Technical Support department of an VoIP-based Contact Center. In May 2010 Miguel joined FCCN, the Foundation for National Scientific Computing, where he is currently working as a VoIP System engineer and provides technical support and management to the VoIP@RCTS project.

Table of Contents

1 IPT Best Practices from Czech Republic

Miroslav VOZNAK

IP telephony, as one of the advanced services offered by the CESNET2 network, attracts more and more users. For this reason, CESNET puts an appropriate emphasis on further development of the IP telephony infrastructure and improving its quality. About one million voice calls through the CESNET2 network were carried out during 2010 [CES2010]. These numbers clearly demonstrate the utility and popularity of this service.

1.1 IP Telephony Infrastructure

CESNET received an additional IP telephony network access prefix 950 0 from the Czech Telecommunication Office in 2004, so 100 thousand numbers are available for IP phones and other clients.

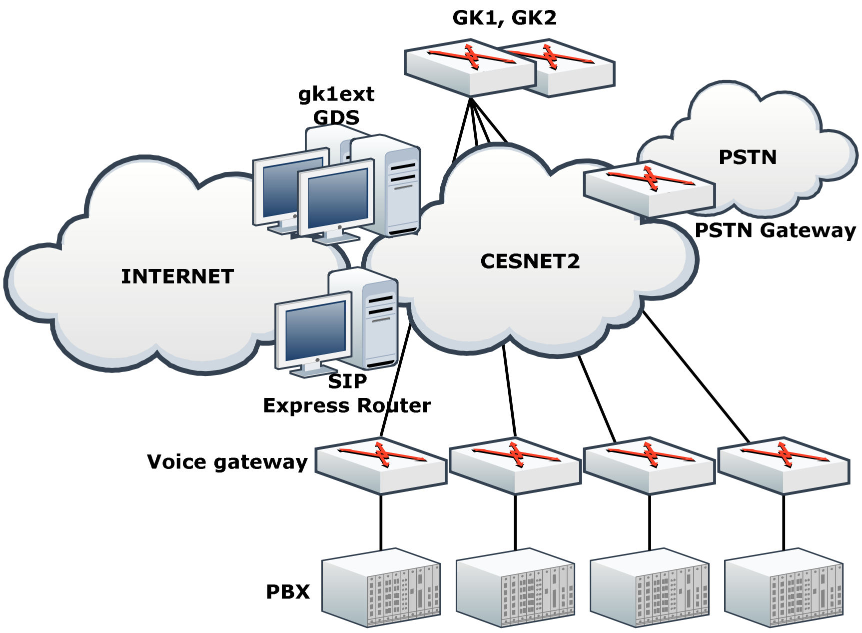

Figure 1. IP telephony infrastructure.

CESNET has been providing IP telephony since 1999 when the first interconnection between four universities was implemented. Subsequently, CESNET built the key elements of H.323 infrastructure with two gatekeepers a Voice gateways. These gateways were based on Cisco routers, with PBX's of CESNET members behind them. Today, there are overall 46 gateways with PBX's of Czech universities.

Figure 1 shows a diagram of the IP telephony infrastructure. Voice gateways of CESNET members are registered to the internal gatekeepers GK1 and GK2. Reachability from the Internet is provided by the border gatekeepers and SER (SIP Express Router). All gateways support bidirectional communication via H.323 and almost all are also able to accept incoming SIP calls. SER ensures handling SIP protocol calls. Connections requiring translation between SIP and H.323 pass through the translation gateway (Cisco IP2IP Gateway), depicted in Figure 2.

1.2 SIP Protocol Implementation

The main control element of the SIP protocol implementation in the CESNET2 network is a SIP proxy based on SIP Express Router (SER). The proxy also functions as the registrar server. Software or hardware SIP clients can register with the proxy and communicate through it. The SIP proxy also routes the calls to the gateways of the connected institutions according to the telephone number prefixes. For gateways that do not support SIP, the calls are routed through a SIP/H.323 gateway. The proxy also handles calls to and from other SIP domains. Calls initiated from the PBXs behind the gateways, from PSTN and from H.323 VoIP clients whose destination is the SIP network also pass through the SIP/H.323 gateway.

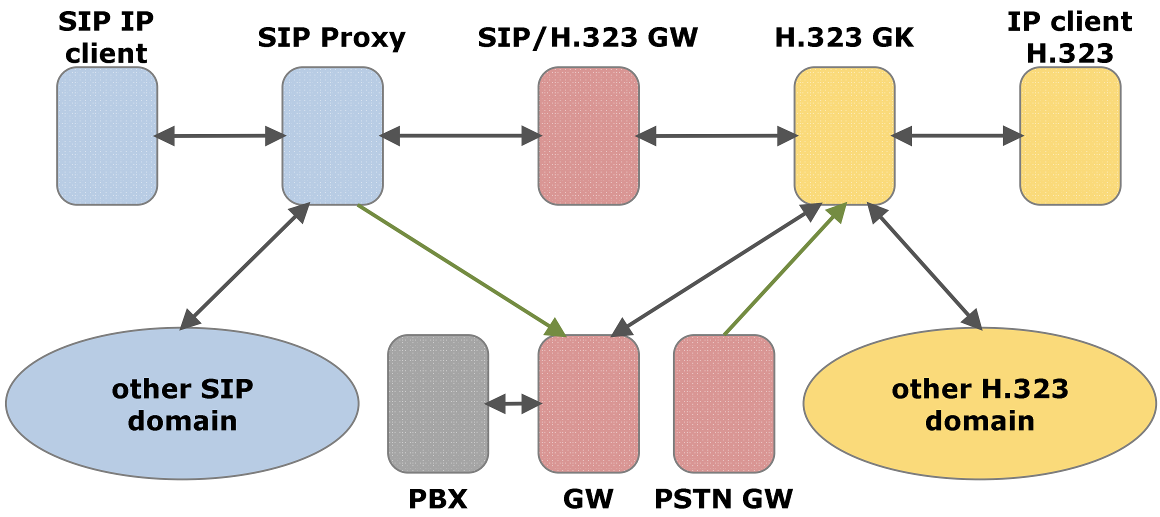

Figure 2. Scheme of the SIP and H.323 network interconnection.

For a domain to work properly with the multidomain SIP proxy, correct routing of requests must be ensured by means of a DNS SRV record in the institution's DNS zone. End users can create their accounts via a web form. Requester's identity is verified using the eduroam infrastructure that uses AA systems of requester's home organisation. In the near future, along with the deployment of a new distributed AAI, we expect even broader and more effective use of AAI, mainly in the area of interdomain authentication and shared service access.

Together with the SIP server we launched a new web site - IpTelWiki - dedicated primarily to the SIP implementation in the CESNET2 network [CESipt]. This website providess information about the protocol itself and and SIP integration into the existing CESNET2 IP telephony network. It also provides a web interface to the SIP proxy (SerWeb) where users can manage their SIP accounts and registrations and also obtain call statistics and other data.

1.3 Best practices of advanced IP telephony at Czech universities

The following subchapters describe several examples of advanced IP telephony best practices. More detailed information is included in the Best Practice Document – Set of implementations in the Czech academic environment [IPTreview].

1.3.1 Auto-Configuration System

IP telephony at the University of West Bohemia (UWB) is based on an open-source solution (Kamailio) and they are using an interesting auto-configuration system (AS). AS enables an automated installation of Linksys IP telephones and allows an installation without taking up administrators' time usually necessary to install such telephones. The whole Auto-configuration System cooperates with an OpenSER which uses a MySQL database to store its configuration. The configurations are distributed through the TFTP protocol and are downloaded by IP telephones when they start up for the first time [petrovic].

1.3.2 Automatic Attendant

An automatic attendant is the result of the research activity at the Department of Cybernetics of the West Bohemia University. It allows callers to be automatically transferred to an extension without the intervention of an operator (typically a receptionist). The Department of Cybernetics applied their own speech recognition algorithms (ASR) to ensure that the called person is recognised and the call transferred to the called party. First version was developed in 2003. In addition to ASR technology, the automatic attendant involves using dialogue system based on VoiceXML, Oracle database and text to speech (TTS) technology. The SIP interoperability of the automatic attendant is ensured by PJSIP open-source client library, the library is a multi-platform one and enables implementing Asterisk in the overall solution.

1.3.3 Accounting from Cisco Gateways

The Czech Technical University (CTU) operates TAS-IP IP telephony accounting application based on data collected from Cisco voice gateways which send information about individual calls through the RADIUS protocol. The billing system is fed call detail records (CDR's) from every single gateway and CDR's are stored in Postgree SQL database. Through RADIUS, CTU collects not only information for billing but also data about the quality of individual calls. Records are imported into an SQL database which serves as the data resource for self-evaluation of the web interface. Cisco gateways evaluate sent Icpif value (Calculated Planning Impairment Factor) calculating estimated speech quality [voz94].

1.3.4 **Monitoring of ENUM Records**

CESNET was very active while ENUM was tested in the Czech Republic. It ensured delegation of appropriate NAPTR records for almost all Czech universities. The fact that an ENUM record exists does not automatically mean that it is available. In this case, CESNET'sENUM monitoring system seems to be useful. Monitoring of ENUM records is based on NAGIOS with check_enum module (plugin) created in PERL. Every prefix is tested using the following procedure: existence in WHOIS database, expiration of validity, availability of DNS server (NS-SET) and availability of SRV records in DNS [IPTreview].

1.3.5 **Protection against Spam over IP telephony**

VSB-Technical University of Ostrava implemented Anti-SPIT application into Asterisk. They focused on Spam over Internet Telephony (SPIT) as a real threat for the future. They have developed both a tool generating SPIT attacks and AntiSPIT tool defending communication systems against SPIT attacks. AntiSPIT represents an effective protection based on statistical blacklist and works without participation of the called party which is its significant advantage. AntiSPIT is able to analyse and process input data from Call Detail Records (CDR's) and consequently determine whether the used source will be inserted into a blacklist. CDR's are an integral part of every PBX and it was decided to implement AntiSPIT also into Asterisk PBX. The application gives an output which is inserted as a command which can control the blacklist. Asterisk provides CLI interface enabling us to create or delete the particular records in the blacklist database [voz175].

1.3.6 **PHP OpenSER Administrator**

The University of Ostrava provides IP telephony for their employees with its own developed user-friendly web interface POSERA (PHP OpenSER Administrator). POSERA was implemented in PHP and enables to set up user accounts in OpenSER through the web (HTTPS). The users are verified through LDAP in a corporate directory and then can fill in a form and the new account in OpenSER is created after confirmation. POSERA enables not only creating SIP accounts but also their administration, such as administration of personal information or displaying missed calls [IPTreview].

2 IPT Best Practices from Finland

Ulf TIGERSTED

In 2007, the Åbo Akademi university in Turku, Finland encountered a major issue. The existing telephone exchange was old, its infrastructure was badly supported and the cost of the rented wires? for the extensions was extremely high. Given the age of the campus in Turku (Åbo in Swedish), individual departments had been scattered in several small buildings along the Biskopsgatan street. By 2007, the humanities department had been moved to a newly renovated facility but still there was a need for 300 extensions out of 1100 to be leased from the main PBX or the three satellite exchanges. With cost of around 25€ per leased line per month, the administration was keen on implementing saving measures. The PBX was a Nortel Meridian 1 Option 61C with three Option 11 satellite exchanges, with the main exchange bought in 1997 and the satellite exchanges bought second hand as they were needed [tiger].

2.1 Introduction

The other two campuses of ÅA, Vaasa and Pietarsaari, were equipped with a modern Siemens HiPath 4000. Pietarsaari had a satellite exchange of the main exchange in Vaasa. The HiPath supported Siemens proprietary VoIP telephones but they were rather basic and expensive.

I got involved in the project when the planning manager of the administration heard I had some experience with the VoIP systems. A working group was formed and assigned the task to analyse the available options and recommend one for further action. The group carried out a survey among the personnel to receive hints about what the personnel needed and expected from their telephones. The result contained no big surprises. A small but very vocal group demanded mobile phones for everyone, as they had departments that could afford to buy them. Most respondents wanted a desk telephone but the majority did not bother to fill in the questionnaire at all. A recommendation to check the actual costs for two different systems was put into the final report: first a VoIP based system utilizing the existing LAN on the campus and the second using mobile phones with a "mobile PBX" that the GSM carriers dearly wanted to sell us.

After this, I got hired as a project engineer to work on the modernization of the Turku PBX with a less costly solution. We talked to the vendors, and heard what they had to offer for a modern VoIP PBX: Cisco, Siemens, Avaya and Alcatel-Lucent all wanted our money. The big GSM carriers also presented us their offers but it was too expensive for the amount of extensions we needed even when not taking the cost of handsets into account.

Another argument against mobile solution was that four graduate students could easily share an office and a desk telephone but to share a single mobile phone would be hard.

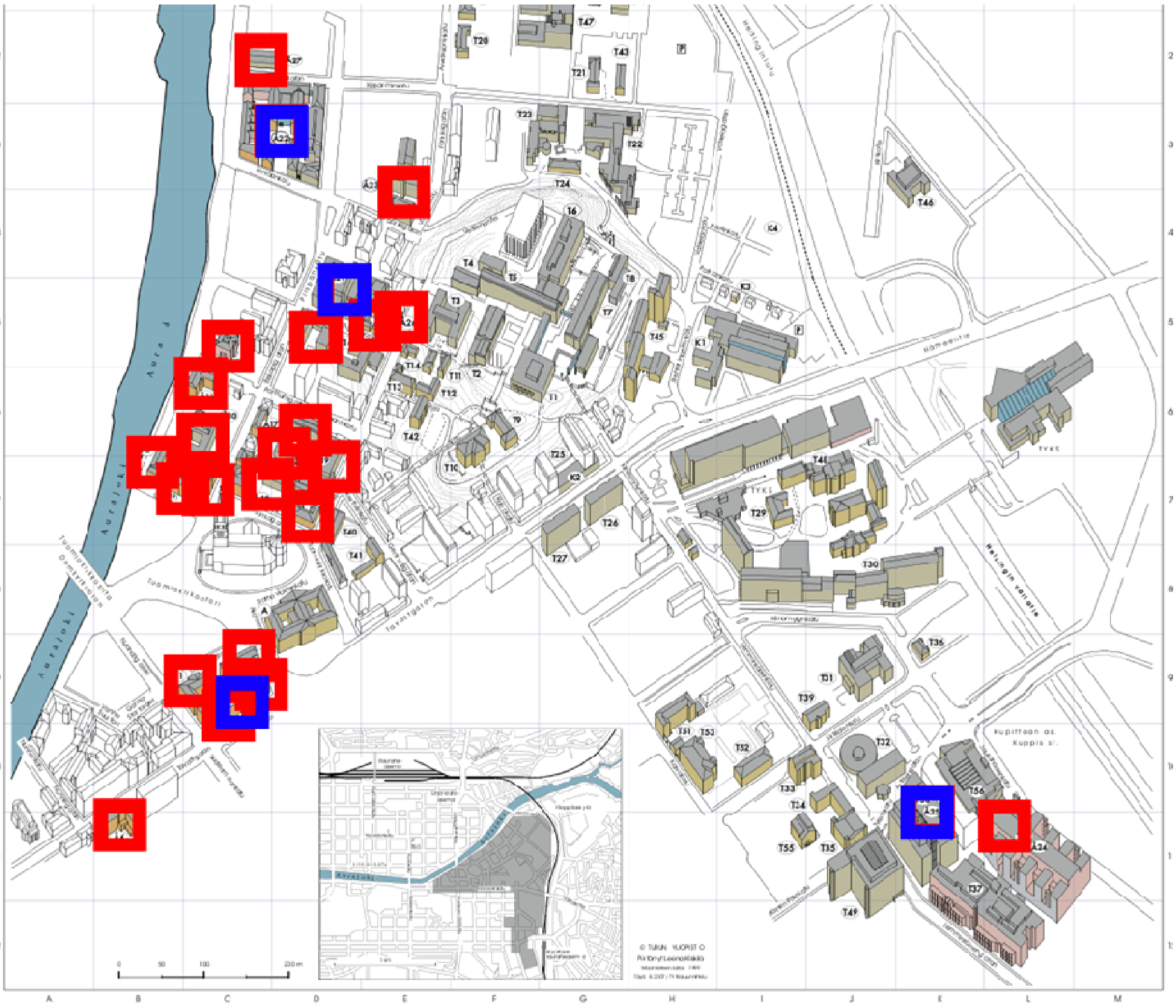

Figure 3. Map of the Åbo Akademi campus in Turku in 2008: Blue squares mark houses with the PBX or satellite exchanges, red squares mark houses with activity.

2.2 Requirement analysis

After reviewing vendor offers, I defined procurement specifications. The final requirements were:

- Least cost wins.
- VoIP desk telephones with PoE possibility and graphical display.
- Standard PC servers as the PBX.
- Usage of a standard VoIP protocol.
- LDAP integration with the exchange or operators interface.
- A duplex system with seamless fail over for the servers and outgoing lines with the possibility to locate the servers in different houses.
- The possibility to use a SIP trunk to connect to the national telephone network and other organizations.

- An optional expansion to Vaasa and Pietarsaari should be taken into account when bidding for the system in Turku.
- The possibility to connect to the PBX with SIP clients.
- Some kind of security, at least for authenticating the telephones to the PBX.

2.3 Implementation and experience

The procurement had to be handled via the Finnish authority for government procurement Hilma, which means that the specifications could not be changed in the middle of the process and that anyone could make a bid.

Åbo Akademi received 11 bids, offering Cisco, Siemens, Nortel, Alcatel-Lucent and Avaya. Avaya was the cheapest and the only one to actually meet the specifications. None of the unsuccesful bidders complained to the court.

We ended up with a bid for 1100 telephones + PoE adapters, two G.650 gateways and a pair of Communication Manager + SIP Enablement Server 1U rack servers and the licenses to make it work.

As the system was delivered and installed for testing on the network, we decided not to attempt to connect the old PBX to the new, and we a new E1 from the telephone company that was oneway instead. Thus, the new PBX could make outgoing calls but all incoming calls went through the old one. The main reason why we opted for this solution was that the Nortel PBX only spoke the DPNSS protocol on E1 lines and we did not want to spend money on changing the old hardware in any way, or pay a large amount of money to get the Avaya to speak, as it is a protocol it would never need in the future.

After some initial testing, the new telephones were distributed to end users shortly before the switchover date. The logistics of this, and getting the telephones connected was a lot more complicated than anyone had imagined. The facilities department that handled the distribution was not prepared to handle several cubic meters of telephones in one week, even with large boxes individually labeled with a department and house. Only one large shipment got lost, it was found after two weeks behind some crates in the library.

Every extension got a unique password and these were distributed to the secretaries of individual department. The “industry standard” was having the same password for all phones, but given the large organization and the variety of users at the university we actually needed some security. At the switchover date, the roles of the PBXes were reversed: The Nortel could only call out and the Avaya could finally receive calls from the outside world. This gave us time to sort out the users that had been forgotten in the first configuration round, and to sort out the faxes. Each gateway had a 24 port analogue line card, so some faxes could be connected directly to it. Elsewhere a decree was given that the limit is one fax per building, and that fax was connected via the university's own line or via an external phone line directly to the telephone company. The same procedure was followed for some alarms (both for circulation pumps, burglary and assault). Most door telephones were connected via ATAs to the SIP server, but the stability of the ATAs was not fully satisfactory as there were issues with stability.

Once everything worked well in Turku, the expansion to Vaasa was ordered. In Vaasa, there was an unused range of numbers (77xx) that got a dual role: The Pietarsaari numbers (previously 50xx) got remapped to 77xx, and 77xx was also configured by the operator company to an inbound range of numbers for the new PBX. Vaasa got an Avaya G450 gateway with a LSP (Local Survivability Processor) feature. This enables it to sustain the whole system’s operation even if both Communication Manager servers in Turku fail. It also has eight analogue ports to connect faxes in. We were afraid of latency issues, as the IP traffic from Turku to Vaasa was routed via Oulu, a round trip of 2000 km. Moreover, the voice traffic is routed to Vaasa inside an OpenVPN layer 2 pipe. We never experienced any latency issues for voice calls, but the soft console software used by the operator in Vaasa was crippled by trying to read the shared contact database one record at a time with locking from a CIFS share, giving a startup time of over 5 minutes.

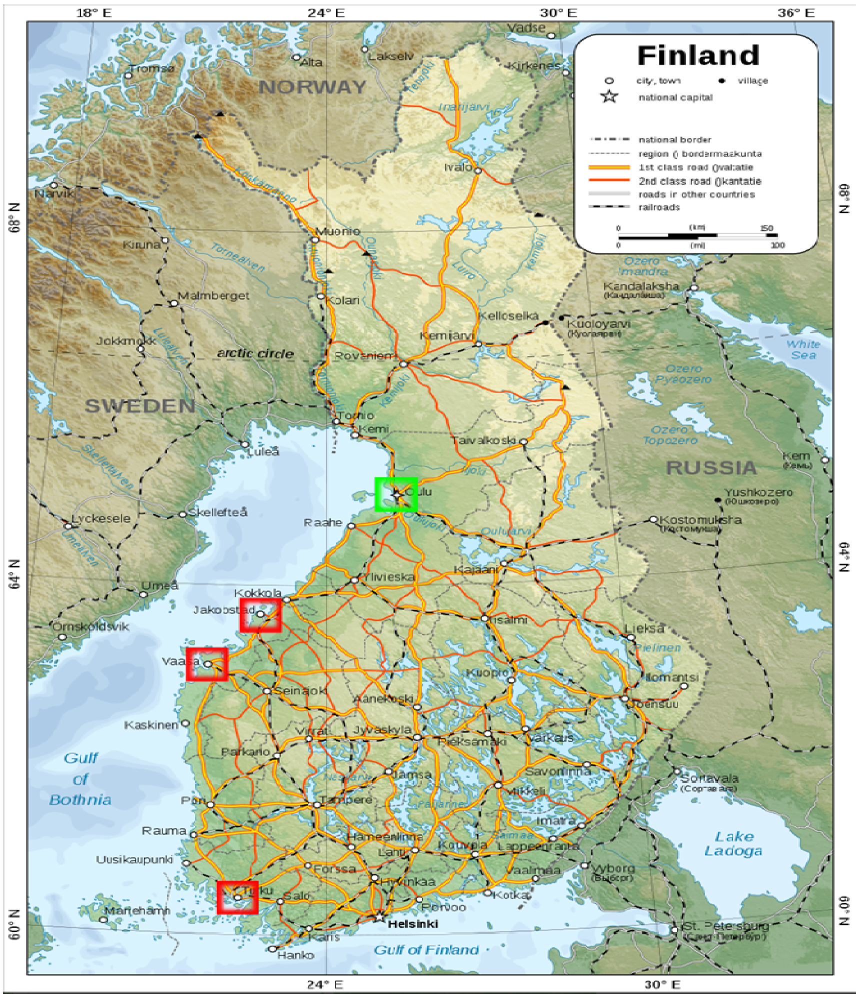

Figure 4. From bottom up: Turku, Vaasa and Pietarsaari (in red) and Oulu (in green) showing the distances.

The final shutdown of the old PBX in Turku was somewhat dramatic. The logs for outgoing calls from the old PBX were not thoroughly examined before shutting it down, and a week later a lone assault alarm was found to still have been connected. After a few days of no heartbeat connection from the alarm to the security center they reported the error in and the forgotten alarm was found and connected to the new PBX. The logs and debit information from the old PBX were handled by an old pair of Pentium/Pentium II computers that had had no maintenance since they were installed. The old PBX completely died when the power was shut down, and trying to restart it later only produced error messages and blinking LEDs.

We never got the LDAP integration to work. The soft console (the program used by the exchange operators) has a LDAP client, but we never got it to work properly with any of our LDAP servers. We tested solutions with both ActiveDirectory and OpenLDAP, but the client was always unhappy with the results and but the users were unsatisfied with the implementation and being failed in user-configuration, so we do not have in use the LDAP integration..

3 IPT Best Practices from Croatia

Branko RADOJEVIC

CARNet, as a Croatian NREN, has been operating a well established high-speed IP network for years that was reserved exclusively for academic and scientific institutions with approximately 300 connections. In 2004, the network was expanded to reach other educational establishments (primary and secondary schools) and today, the number of connection points is around 3000 [radoj].

3.1 voopIX

CARNet is one of few NRENs in Europe that does not only provide connection to GÉANT and commodity Internet, but also provides a number of services on top of the network infrastructure. These range from videoconferencing to hosting several services in CARNet data centers.

One of many services that run in our network is called voopIX, the goal of which is to provide free telephony service between CARNet member institutions by using Voice over IP standardized protocols and open source solutions. The main goal of this project is to interconnect existing legacy PBXes on institution premises by using VoIP gateways transforming traditional telephony standards (for example ISDN PRI connections) to VoIP. The entire VoopIX project has been developed and maintained by CARNets employees only. Since the project has been in operation for almost three years now, preceded by additional two years of testing, and has proved to be an excellent solution in our environment, we are currently looking into ways to implement even more new services and benefits.

Currently, our reports show that despite the early stage we can terminate around 4% of all calls from institutions that are now using our system. We predict that this will grow to 10-15% of all calls and that should cut costs by about EUR 2 500 000 a year in Croatia.

3.2 Challenges in building national VoIP Network

This project is somewhat evolutionary. It all started about four years ago with just two institutions and seven different locations trying to establish VoIP interconnectivity. Since this initial solution proved to be technically easy to implement and its quality was excellent, we tried to expand it to cover other member institutions.

The first challenge was to build a resilient core network that can accept hundreds of parallel telephone calls through the network. That was achieved in 2005 when we installed five core servers and put the first VoIP backbone into operation.

Another key challenge was to build a custom made VoIP gateway appliance that can suit our needs on VoIP side and at the same time enable connecting legacy PBXs. It took us a year to build such an appliance, and we developed it entirely inside CARNet. The **voopIX Appliance** has several features that are not available in standard equipment available around the globe. For example, there is a built-in failover switch that can bypass our appliance and reconnect PBX back to Telco.

Other challenges include management, security, fault and performance monitoring issues for the whole network, user web interface for users from institutions, etc.

3.3 Where we are now and where we want to go

When drafting this document, we have more than seventy largest institutions in Croatia connected in. These include the Ministry of Science, Education and Sports, several agencies and universities and some of the largest faculties in Croatia. It may seem that this is still far away from the number of 3000 locations but as I said before we focus on the largest institutions with largest telephony installations in this project phase. We hope to be able to expand our VoIP network at least three or four times in year 2011, and are still working on a smaller appliance for smaller institutions (for example small schools on islands).

Figure 5. Current voopIX members as of December 2010.

In addition, we started connecting in several institutions in Europe and the US and we hope eventually to create a global network interconnecting academic, scientific and educational institutions. We have already made some improvements to our core network in order to be able to handle additional load from our foreign partners.

3.4 Technical Overview

At the very beginning, we decided to build the entire network around open source software. As for the VoIP part, we had abundant experience at that time with Asterisk IP PBX, so we decided to use this software for the entire network. This decision may seem a little odd at this time as Asterisk is rarely used in such a configuration but it proved to be a good option for us.

First of all, we built our core network which is distributed across Croatia, at this time consisting of five servers running Asterisk. Those five servers do have full-mesh logical interconnection between them with DUNDi protocol running on top of that. One server can be faulty at any time without service disruption.

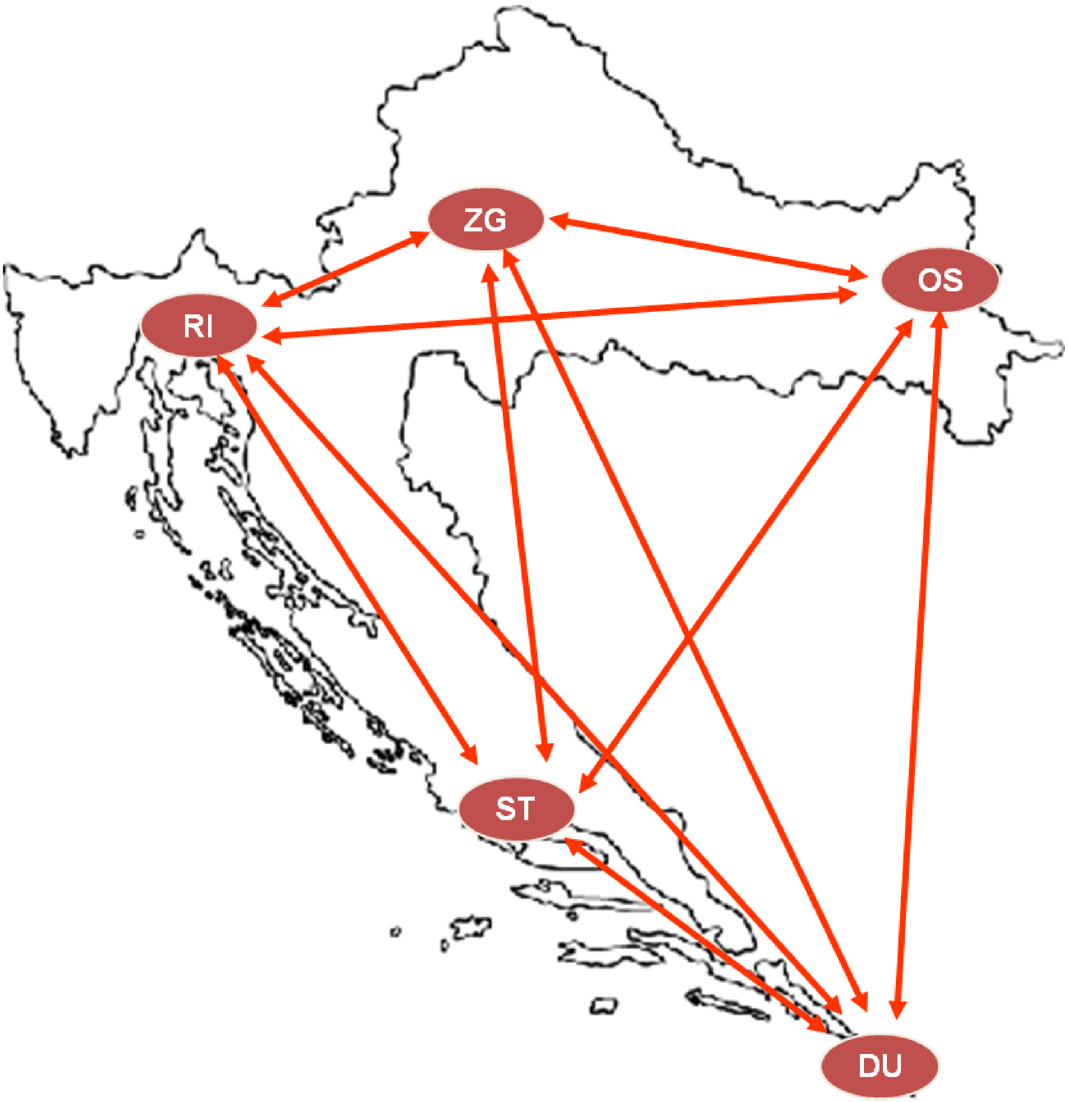

Figure 6. VoopIX core servers in logical full-mesh DUNDi peering as of December, 2010.

On the customer premises at CARNets members institutions, we deploy our voopIX appliance which is actually a small diskless Linux server which is also running Asterisk. This appliance is positioned between legacy PBX and telco lines (usually ISDN PRI, but can be others too) where it “intercepts” calls. Those calls that are available in our system are re-routed through voopIX and all others are connected through the telco lines.

When the appliance first starts, it registers with two of the neighboring core servers by using IAX2 registration. When it becomes available in the IAX2 registration table, then its numbers are published in the DUNDi cloud. We have measured the length of the transitions from registered to unregistered status and backwards. They were done in less than half a second, which ensures that we have a highly dynamical cloud which can become up-to-day almost instantly. Since we have our own appliance and do not use simple VoIP gateways only, we can deploy features that could not be deployed in other situations. Those include additional features that are not currently available (like voice menus, call recording + delivery via email, etc.) on the legacy PBX, or may be a rather expensive upgrade to aged equipment.

We also support “Skype-to-legacy PBX” calls which are all routed through our infrastructure. We have enabled this feature by using Skype for Asterisk drivers, and these are in fact the only licenses that we purchased for the project.

At the end we developed an ENUM-to-DUNDI gateway which allows us to bridge these two different VoIP worlds and enable making calls in both directions.

3.5 Conclusion

Since the inception of this project, we have opened a new page with new possibilities that can be included in our project almost every month. We are in touch with all our partners in Croatia and abroad, to keep the ideas flowing in both directions. We are looking forward to any opportunity to share our knowledge with others.

4 VoIP Practice in Slovakia

Michal HALAS

In Slovakia, the Slovak University of Technology in Bratislava is the best example in the academic sphere. Its Department of Telecommunications implemented the Alcatel-Lucent OmniPCX Enterprise system.

4.1 VoIP in real use

IP telephones together withanalogue and ISDN telephones are connected to the system as depicted in Figure 7. IP telephones are powered solely from PoE network switches.

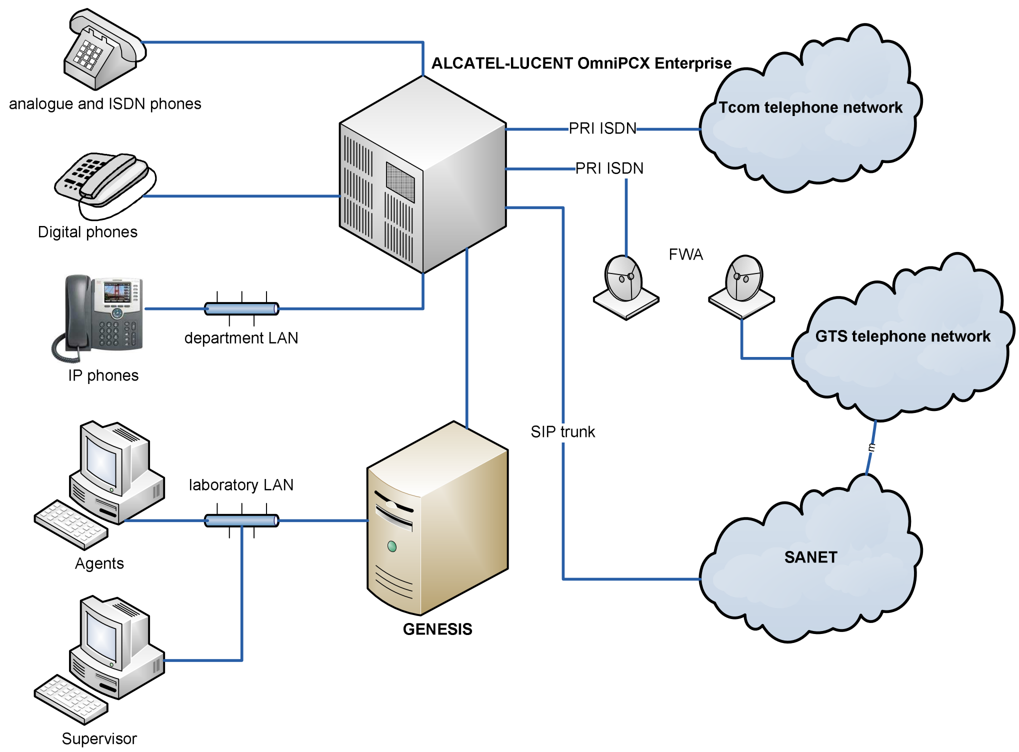

Figure 7. VoIP network scheme.

The system is connected to the public telephone network using PRA ISDN access via the SDH network to the T-Com telephone network.

Another PRA ISDN access uses FWA technology to connect the system to the GTS network. Moreover, the system is also connected to the GTS telephone network through the SANET network and a SIP trunk. Finally, the SANET network also connects the system to several universities in Slovakia and the Czech Republic, again through IP SIP trunks.

A GENESIS contact centre, used for research purposes and to support the education process, is attached to the Alcatel-Lucent OmniPCX Enterprise system Supervisor and agent workplaces are connected to the contact central through the LAN network. The Genesis Contact Centre solution includes sophisticated call routing, comprehensive contact management, robust e-mail management, chat and Web collaboration and outbound dialling capabilities.

4.2 **VoIP in Education**

An experimental lab equipped with multiple Asterisk and openSIPS servers was built for education purposes. . During their studies, students can experiment with several different services and technologies.

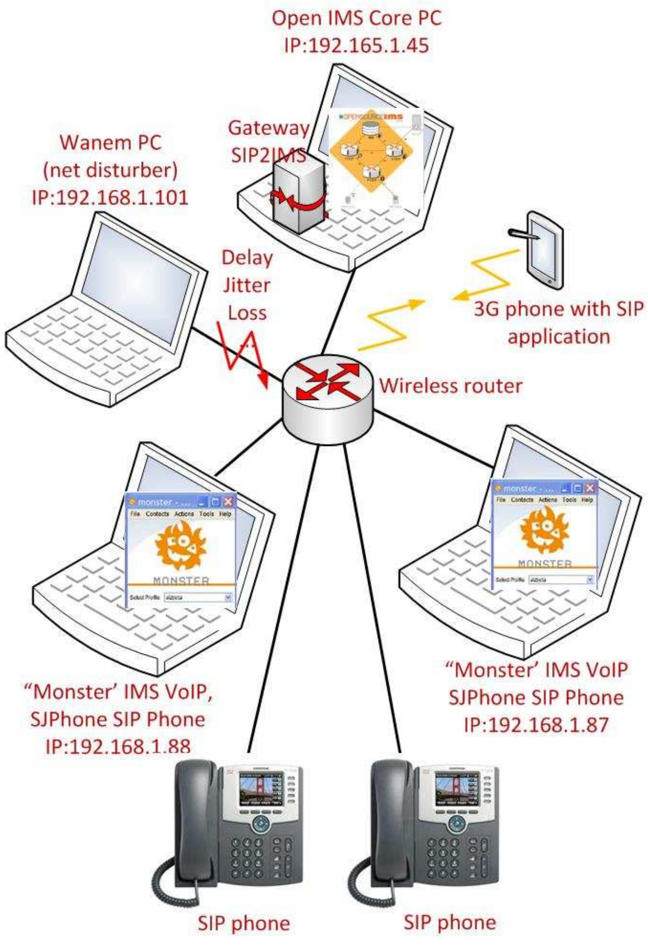

Figure 8. OpenIMS laboratory's structure.

The IMS test facility based on the OpenIMS project was designed for research activity. The project aims at creating an IMS environment for testing QoS of VoIP services in IMS, as well as for testing the interconnection between OpenIMS and Open Source PBX, connection of SoftVoIP applications and VoIP telephones to IMS network through the SIP2IMS gates.

During student projects in the laboratory, students familiarize themselves with the protocols in IMS and VoIP technology, VoIP quality of service in the IMS. The OpenIMS laboratory allows students to set up different network configurations and monitor their impact on the quality of communication.

4.3 VoIP in research Activities

Another laboratory on the basis of IMS Enterprise products, this time with the Alcatel-Lucent's IP Multimedia Subsystem, was built to support research of QoS and services in the IMS network This laboratory was built under the Research and Development Excellence Center - SMART technology Operational Programme (Platform for IMS NGN) financed by the European Union.

The core network is built on the Alcatel-Lucent 5060 IP Call Server supplemented by the access and backbone networks based on IP/ MPLS technology. The IMS infrastructure also covers a management network and multimedia management subsystem.

5 VoIP@RCTS The Portuguese Academic Voice Private

Miguel DUARTE

The VoIP@RCTS project started around 2007 following the network infrastructure upgrade of Rede Ciência, Tecnologia e Sociedade (RCTS) – the Portuguese NREN – that connected around 85% of the Portuguese research and higher education community over a dedicated optical fiber network. VoIP over RCTS was the next logical step in order to exploit all the bandwidth that became available [mouta].

5.1 Introduction

The main motivation, besides the obvious and omnipresent need to lower operating costs, was to provide higher education institutions the needed infrastructure to transport voice traffic over IP in a converged, integrated and safe manner.

In order to accomplish this goal, FCCN – the Foundation for National Scientific Computing – organised two separated tenders. The first tender aimed at the acquisition of equipment needed to converge the existing legacy telephony systems with a new VoIP infrastructure on each of 43 higher education and research institutions in Portugal. For each institution, one media gateway per legacy PBX, at least one IP PBX, one dedicated server for the purpose of accounting, billing and QoS report, and one Session Border Controller to close and manage the institution VoIP private network were purchased. At this stage, it was possible to implement a free of charge call routing mechanism across all the VoIP@RCTS community. FCCN implemented the ENUM service using 1.3.5.nrenum.net delegation. This service enables an end-to-end call routing over the RCTS network, interconnecting all these institutions without using any Telco's infrastructure.

The second tender aimed at the procurement of public voice services for these participants. The new voice services were acquired in order to connect the newly implemented VoIP systems to the fixed and mobile voice public networks over IP. These new services resulted in the implementation of SIP trunking between the four major Telco's in Portugal and each participant institution's Session Border Controller and backup ISDN lines connected with each media gateway.

This VoIP infrastructure had been, and has been since, implemented gradually mostly due to the unavoidable need to integrate all traditional PBX systems in order to circumvent the infeasible scenario of having to abruptly purchase and install an entire new VoIP telephone infrastructure. Using this solution enabled to maintain the legacy telephone systems in use, and plan investments for a smooth migration to fully VoIP solutions each at their own pace, while at the same time guaranteeing that all inter-institutions voice traffic flow over RCTS.

Nowadays VoIP@RCTS can already be described as a nation-wide virtual private voice network composed of 43 institutions with more than 35.000 DIDs, directly interconnected via ENUM and via SIP trunks with the four major Telco's in Portugal as well as via TDM circuits in a resilient way.

5.2 **Technical Architecture**

Each institution has its own private voice network that is composed of the following network elements:

The Media Gateway (MGW) – is responsible for the interconnection between the PSTN and VoIP worlds. It processes all the signalling and media-related translations and transcoding involved in establishing a call between the two systems. Basically, the MGW establishes a bridge between the legacy PBXs and public switched telephone network (PSTN) with the VoIP infrastructure.

The interconnection with the PSTN is established using ISDN circuits (BRI and PRI). In the VoIP@RCTS model, these circuits should only be used for FAX, ATM payment terminals and service monitoring reporting (alarms) connections, and as redundant voice circuits in case of any main VoIP trunk's failure.

The iPBX is the IP PBX (telephone system) and central point of each institution's voice service configuration. It is responsible for SIP accounts management, call routing and voice service functionalities as ring groups, interactive voice response (IVR), direct inward system access (DISA), conferencing, voicemail, etc.

Within the VoIP core infrastructure, the iPBX acts as a SIP Core Proxy, SIP Register Server and Application Server and it is a SIP and IAX2 compliant system.

The Session Border Controller (SBC) acts as a VoIP border gateway separating and controlling all the institution's external VoIP traffic (inbound and outbound). It can be separated into two logical functions: Signalling, where it controls and filters all the SIP signalling that crosses the border of each institution's voice private network and also manipulates the contents of these messages; and Media that controls the media streams exchanged with exterior networks, provides QoS and prevents DoS attacks.

Finally, the Accounting Billing and QoS Report Server (SABQR) gathers all the information required for each institution's voice service, namely accounting, billing and QoS metrics. Enables call control and monitoring, costs analysis and call reporting on the overall institution's VoIP traffic.

These network elements are distributed per institution using the following logic: one MGW installed per department (organic unit) and installed in front of every legacy PBX in cases where one is still being used. One iPBX where all the SIP users are provisioned, the voice services configured and where all the SIP terminals are registered. One SBC that protects and handles all institution's VoIP traffic to and from the media gateways and iPBX. Additionally there is also a generic server where there is project customized software for Accounting Billing and QoS Report (SABQR) installed for the overall institution's voice traffic.

Another VoIP@RCTS project goal was to provide unified communications federated software to the national research and educational community over the implemented VoIP architecture. This federated software named "Arara" consists of a softphone based on technologies like XMPP, SIP, and SAML2 and it provides the certified community an opportunity to access the SIP world. Through this federated softphone, it is possible to use institutional SIP accounts anytime, anywhere. This open standard tool also enables video conference calls, presence and instant messaging (IM).

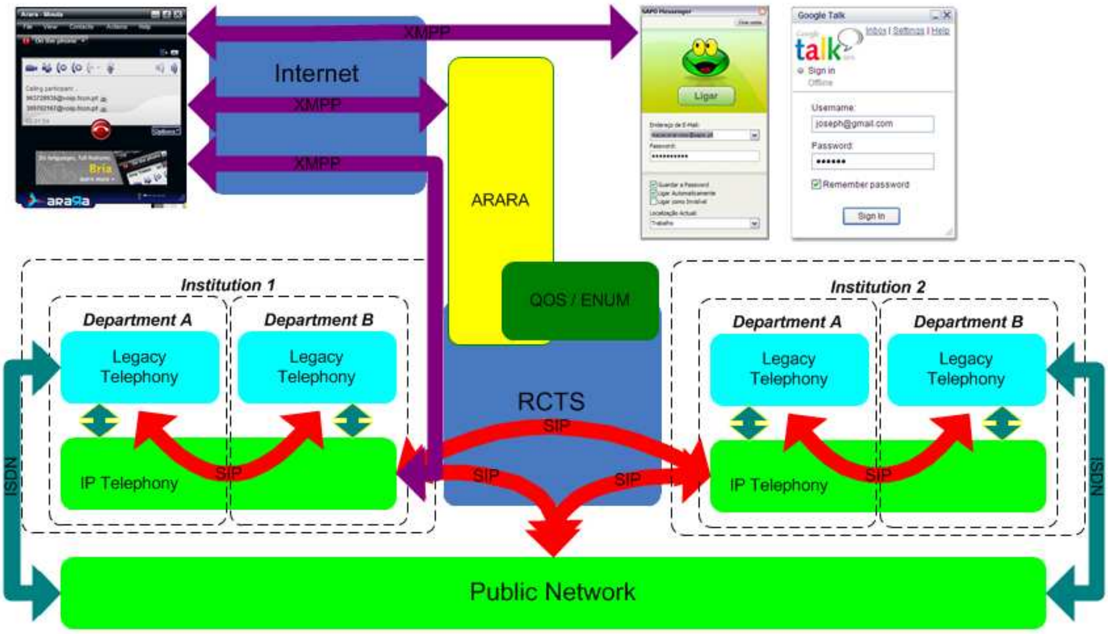

Figure 9. Arara – Federated VoIP@RCTS Softclient.

In terms of IPv6 interoperability, the VoIP@RCTS project is still in an early development stage. Although the FCCN's voice private network is still implemented over IPv4, there is already a corporate laptop IPv6 bridging over the installed SIP hardphones. All corporate laptops are currently running on IPv6 while the hardphones are still using IPv4.

FCCN is, however,already looking ahead and planning on an overall IPv6 VoIP infrastructure to be implemented in the near future.

In respect to system monitoring and considering VoIP@RCTS characteristics and wide infrastructure, there was a need to implement a central monitoring platform that guarantees an efficient detection of voice service downtimes caused by infrastructure anomalies such as existing network problems, SW or HW deficiencies and ISDN circuit failures. The implemented system not only allows the monitoring of general Server and Linux metrics but also specific VoIP services like SIP Trunks, PRI and BRI lines and active calls.

This platform triggers error description emails to each institution's IT department in case an error is detected.

5.3 ENUM as Routing Service

To implement a free of charge VoIP call routing over the RTCS network between all the institutions involved in the project, FCCN implemented the ENUM service based on TERENA's delegation[1] of 1.5.3.nrenum.net. According to the proposed architecture, the media gateway or iPBX, depending on the origin of the call, is responsible to perform the ENUM queries and, based on the given answers, route the calls to any destination: ENUM destination, Telco's SIP Trunk or failover through TDM backup circuits in case of IP connectivity failure, SBC malfunction or problems related to the Telco Operators VoIP infrastructure.

Prior to the ENUM call establishment, the system performs three ENUM queries using the following order: *e164.arpa*, *nrenum.net*, *e164.org*. If any of the previous queries is answered positively, the call is routed over the RCTS/internet to its destination SIP peer.

With the successful implementation of an ENUM standard and SIP compliant solution, it was imperative to extend FCCN's VoIP horizons to other European congeners. Accordingly, FCCN already established two VoIP peering connections with the following NREN's: NIIF (Hungary) and Carnet (Croatia).

This type of peering allows a free establishment of international calls between NREN's via ENUM.

One of FCCN goals for 2011 is to spread our international VoIP peering interconnectivity to include the rest of the NREN's.

5.4 Telco Operators Connections

To accomplish one of the main goals of the project – to promote a converged aggregate of reliable VoIP services between all the institutions and Telco's Operators in the market, through RCTS network – FCCN carried out a public tender to acquire new voice services for all the involved institutions. This new service implemented between each institution and the four major Telco's in Portugal is composed by dedicated SIP trunks with Portugal Telecom (PT Prime), TMN, Vodafone and Sonaecom.

For a matter of resilience and also as a part of the PT Prime (Fixed line telco operator) contract, a TDM circuit was installed in each media gateway. This ensures maximum voice service availability in case of failure with call completion over VoIP (ENUM or SIP trunk) – Crankback mechanism. This mechanism basically guarantees voice service redundancy and work as follows: if something goes wrong in the IP world, both infrastructures – carrier and institution's voice private network – automatically re-route the calls via TDM circuits.

Besides ensuring high voice service availability, these "backup" circuits also cover the need to establish emergency calls, FAX and ATM payment terminals over TDM.

[1] For more information about FCCN's nrenum.net involvement, consult the following link: https://www.nrenum.net/index.php/Main_Page

5.5 Conclusion

In such a wide project, spread across the country and involving the portability of 35.000 DIDs within 220 media gateways and 80 iPBXs, around fifty per cent of the work is organizational work. It is essential to keep in mind that technical and non-technical information about every site should be organized in a structured way and should be easy to access. This data will be consulted many times by different users and there will be a constant flow of additional information to be added during the rollout.

Along the project, customization at different levels was the key factor, namely: in media gateway euroISDN stacks to ensure the proper interoperability with all the involved legacy PBXs, in media gateway and SBCs dial plans to ensure routing specific data and fax calls through TDM circuits, call detail record fields generated on each VoIP network element, etc. In what concerns to customization and integration, the usage of Open Source software, Asterisk and OpenSER, in the media gateways proved a real asset.

There is also the need to establish solid and well-structured support procedures to account for all the service and infrastructure maintenance as well as preventive actions that can minimize any possible service outages.

As we are dealing with state of the art technology there is also a need to invest some effort to share all necessary information related to the project and used technology with all parties involved in order to provide them with a comfortable and sustainable level of management infrastructure independency. This is being delivered by FCCN through workshops and wikis accessible to all the VoIP community.

Legal regulation for SIP trunks, geographical number usage and emergency (112) call location are still under development. Consequently, it is very important to predefine, as much as possible and contractually, all requirements in what concerns service level agreement, QoS metrics, call detail record fields available and numbering plan to use. There is always space to improve the resilience of such a large and heterogeneous solution. Currently, the project is in its fine-tuning and improvement stage.

Communication and cooperation between ANACOM, FCCN's VoIP engineering team and FCCN's legal department was a cornerstone in the project's success.

Analyzing the project in the long term run, we do believe that by installing 220 media gateways, 80 iPBXs and 43 Session Border Controllers properly integrated on one side and the entire legacy PBXs involved and on the other side with Telco's via SIP Trunks and ENUM, we took the first step to unchain these institutions from the proprietary and closed market of legacy PBXs to the new world of SIP and Open Standards.

Analyzing the project from an economic point of view and trying to measure the immediate impact of this project in terms of earnings and cost savings, we can say that globally 7-10% of voice calls become free of charge thanks to ENUM service; and total telecommunications costs decrease by 48% per participant on average thanks to the new voice services acquired in the public tender.

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Glossary

AA	Authentication and Authorization
ASR	Authomatic Speech Recognition
BRI	Basic Rate Interface
CDR	Call Detail Record
DISA	Direct Inward System Access
DNS	Domain Name Service
DPNSS	Digital Private Network Signalling System
DUNDI	Distributed Universal Number Discovery
ENUM	E.164 Number Mapping
FCCN	Foundation for National Scientific Computing
GK	Gatekeeper (H.323 network key element)
GW	Gateway
HTTPS	Hypertext Transfer Protocol Secure
IAX	InterAsterisk Exchange
IMS	IP Multimedia Subsystem
ISDN	Integrated Services Digital Network
IVR	Interactive Voice Response
LAN	Local Area Network
LDAP	Lightweight Directory Access Protocol
MGW	Media Gateway
NAPTR	Name Authority Pointer
NGN	Next Generation Network
NREN	National Research and Education Network
PBX	Public Exchange
PHP	PHP H\ypertext Preprocessor
PoE	Power over Ethernet
PRA	Primary Rate Access
PRI	Primary Rate Interface
PSTN	Pblic Switched Telephone Network
QoS	Quality of Service
RADIUS	Remote Authentication Dial In User Service
RCTS	Rede nacional de investigação e ensino (NREN)
SBC	Session Border Controller
SDH	Synchronous Digital Hierarchy

SER	SIP Express Router
SIP	Session Initiation Protocol
SPIT	Spam over IP Telephony
SQL	Structured Query Language
TLS	Transport Layer Security
TTS	Text to Speech
XMPP	Extensible Messaging and Presence Protocol