VoIP Network Status in Portuguese R&E

Best Practice Document

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Authors: N. Gonçalves (FCCN), C. Friaças (FCCN)

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Contact:	nuno@fccn.pt

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Table of Contents

Executive Summary

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 fibre network. The remaining 15% of this community is also enabled, but over leased connections. VoIP over RCTS was the next logical step in order to exploit all the bandwidth that became available.

In late 2012, the Portuguese government decided to integrate FCCN, the organisation managing the NREN into Fundação para a Ciência e a Tecnologia, I.P. (FCT,I.P.). That integration process was completed in October 2013. FCT, I.P. is the public institute responsible for funding science in Portugal, and FCCN became one of its units, still with the main responsibility of managing RCTS.

This document is essentially an update on the Portuguese chapter of CBPD146, published during the GN3 project [**1**].

1 Introduction

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

In order to accomplish this goal, FCT organised two separated tenders. The first tender aimed at the acquisition of equipment needed to achieve convergence between the existing legacy telephony systems and a new VoIP infrastructure on each of 43 higher education and research institutions in Portugal back then. 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 reporting, and one Session Border Controller to close and manage the institution's 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. FCT 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 Telcos in Portugal (Portugal Telecom, MEO, Vodafone and NOS) and each participant institution's Session Border Controller and backup ISDN lines connected with each media gateway.

This VoIP infrastructure was, and has been since, implemented gradually, mainly due to the unavoidable need to integrate all traditional PBX systems in order to avoid having to abruptly purchase and install an entire new VoIP telephone infrastructure. Using this solution enabled all participating organisations 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-institutional voice traffic was flowing over RCTS.

Nowadays, VoIP@RCTS can already be described as a resilient nationwide virtual private voice network composed of 37 institutions with more than 45,000 DIDs, directly interconnected via ENUM and via SIP trunks with the four major Telcos in Portugal, as well as via TDM circuits.

2 Technical Architecture

Each institution connected to RCTS 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 such 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 the entire 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 Server gathers all the information required for each institution’s voice service, namely accounting and billing. It 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 is installed per department (organic unit) and is 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 customised software for Accounting/Billing installed for the overall institution's voice traffic.

In terms of IPv6 interoperability, the VoIP@RCTS project is still in a development stage. Although FCT's voice private network is still implemented over IPv4, there are already corporate laptops connected to the IPv6 network, bridging over the installed SIP hardphones. All corporate laptops are currently running on IPv6 while the hardphones are still exclusively using IPv4. FCT is, however, planning for 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 span, 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, software or hardware 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 departments in case an error is detected.

3 ENUM as Routing Service

To implement free-of-charge VoIP call routing over the RTCS network between all the institutions involved in the project, FCT implemented the ENUM service based on TERENA's delegation of 1.5.3.nrenum.net [**2**].

According to the proposed architecture and depending on the call's origin, either the media gateway or iPBX is responsible for performing ENUM queries. Based on the answers provided, calls are routed to the ENUM destination, the Telco's SIP Trunk or they failover through TDM backup circuits in the 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 network or the Internet to its destination SIP peer.

With the successful implementation of an ENUM standard and an SIP-compliant solution, it was imperative to extend FCT's VoIP horizons to other European counterparts. Accordingly, FCT has already established peerings with other NRENs in the scope of the nrenum.net domain. This type of peering allows a free establishment of international calls between NRENs all over the world via ENUM.

The following figure shows a snapshot of the state of ENUM's usage and peerings in the scope of nrenum.net (source: https://crawler.nrenum.net/).

top country codes

#	country name	E.164	ENUMs
1.)	Hungary	+36	58571
2.)	Portugal	+351	45655
3.)	Switzerland	+41	31301
4.)	Croatia	+385	22727
5.)	Brazil	+55	10128
6.)	Spain	+34	6904
7.)	North American Numbering Plan	+1	5112
8.)	Argentina	+54	3665
9.)	Italy	+39	3236
10.)	Australia	+61	1896
11.)	Greece	+30	911
12.)	United Kingdom	+44	49
13.)	Netherlands	+31	36
14.)	New Zealand	+64	31
15.)	India	+91	22
16.)	Latvia	+371	21
17.)	Hong Kong	+852	15
18.)	Belgium	+32	11
19.)	France	+33	8
20.)	Peru	+51	6
21.)	Colombia	+57	2
22.)	Romania	+40	1
23.)	Poland	+48	1

4 Telco Operators Connections

To accomplish one of the project's main goals – to promote a converged aggregation of reliable VoIP services between all the project member institutions and Telco Operators in the market, through the RCTS network – FCT 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 Telcos in Portugal is composed by dedicated SIP trunks with Portugal Telecom (PT), MEO, Vodafone and NOS.

As a matter of resilience and also as a part of the PT (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 works as follows: if something goes wrong in the IP world, both infrastructures – carrier and institution's voice private network – automatically reroute 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.

5 Conclusion

In such a wide project, spread across the country (Continental Portugal plus the islands of Madeira and the Azores) and involving the portability of 45,655 DIDs within 220 media gateways and 80 iPBXs, around fifty percent of the work is organisational work. It is essential to keep in mind that technical and non-technical information about every site should be organised 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, customisation 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 respect of customisation and integration, the use of Open Source software, Asterisk and Kamailio, in the media gateways proved to be 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 minimise 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 FCT through periodic workshops and Wikis accessible to the whole VoIP community.

Legal regulation for SIP trunks, geographical number usage and emergency (112|911) 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, FCT's VoIP engineering team and FCT's legal department was a cornerstone in the project's success.

Analysing the project over the long term, we believe that by installing 220 media gateways, 80 iPBXs and 45 Session Border Controllers properly integrated on one side and the entire legacy PBXs involved and on the other side with Telcos via SIP Trunks and ENUM, FCCN took the first steps in unchaining these institutions from the proprietary and closed market of legacy PBXs to the new world of SIP and Open Standards.

Analysing 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 the ENUM service; and total telecommunications costs decreased by 48% per participant on average, thanks to the new voice services acquired in the public tender between 2009 and 2012. Additionally, another decrease by 50% was assured with the contracts coming up for renewal in the period 2012 to 2015.

References

[1] Best Practice Document: Set of IP Telephony Best Practices in National Research Networks in EU.
Miroslav Voznak, et al, March 2011 (CBPD146, Czech Republic)
http://www.terena.org/activities/campus-bp/pdf/gn3-na3-t4-cbpd146.pdf

[**2**] For information about FCT's nrenum.net involvement, see:
http://www.nrenum.net

Glossary

ATM	Automated Teller Machine
BRI	Basic Rate Interface
DID	Direct Inward Dialling
DISA	Direct Inward System Access
DoS	Denial of Service
ENUM	E.164 Number Mapping
FCCN	Fundação para a Computação Científica Nacional
FCT	Fundação para a Ciência e a Tecnologia, I.P.
IAX2	Inter Asterisk eXchange version 2
IP	Internet Protocol
IPv6	Internet Protocol version 6
ISDN	Integrated Services for Digital Network
IT	Information Technology
IVR	Interactive Voice Response
MGW	Media Gateway
NREN	National Research & Education Network
PBX	Private Branch Exchange
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RCTS	Rede Ciência Tecnologia e Sociedade
SBC	Session Border Controller
SIP	Session Initiation Protocol
TDM	Time Division Multiplexing
TERENA	Trans-European Research and Education Networking Association (since October 2014 the GÉANT Association)
VoIP	Voice over IP