



IST-026546

**Organising Caucasus and Central Asian Internet Offerings to NRENs
OCCASION**

Deliverable D4-1: Voice over IP Facilities for OCCASION

Contractual Date of Delivery to the European Commission:	February 2006
Actual Date of Delivery to the European Commission:	July 2006
Editor(s):	Peter Kirstein
Participant(s):	Peter Kirstein, Kars Ohrenberg, Hans Frese, Piers O'Hanlon, John Andrews
Workpackage :	4
Title of Deliverable:	Provision of facilities of VoIP for the OCCASION community
Security:	Pub
Nature:	O
Version:	Final
Number of pages:	9

Abstract: The Silk Project provides connectivity to GEANT2 for the NRENs of several Newly Independent States of the Caucasus and Central Asia. The OCCASION Project provides Value Added Services over the Silk infrastructure for the benefit of the users. This Deliverable describes the Voice over IP facilities which have been made available by the end of July 2006.

Keywords: Silk Project, NATO, IST, Internet Access, Caucasus, Central Asia, Satellite, VoIP

Project Manager: Professor Peter T Kirstein
Department of Computer Science
University College London, Gower Street
LONDON, WC1E 6BT, U.K.

Phone: +44 (0) 20 7679 7286
Fax: +44 (0) 20 7387 1397
Email: p.kirstein@cs.ucl.ac.uk

Date: **July 31, 2006**

Table of Contents

1. Introduction	3
2. The Cisco-based System	4
2.1. Overview of system architecture.....	4
2.2. Brief overview of SCCP.....	4
2.3. Audio Conferencing	5
3. The Implementation of VoIP for the OCCASION Project	5
3.1. Introduction	5
3.2. Call Managers.....	5
3.3. PSTN Integration.....	5
3.4. Clients	6
3.5. Security	6
3.6. Quality of Service.....	6
4. Extensions Enabled for OCCASION Purposes	6
5. Experience So Far.....	7
6. Regulatory Considerations	8
7. Further Developments Planned.....	9
References.....	9

SUMMARY

This report describes the facilities that have been put in place during the first period of the OCCASION project to provide Voice/IP for the OCCASION community. Under SPONGE, there had been experimentation with DESY running services based on the Cisco proprietary Skinny protocol, and UCL concentrating on the Standard SIP ones for service provision. For the first phase of the OCCASION project we have concentrated on deploying only Skinny – but in a way that would ease deployment for a larger number of telephone extensions. This includes providing Call Managers initially at two sites – but in a way that could be extended if necessary. We have provided audio-conferencing facilities, so that audio-conferences with up to about 20 participants could be held. We have introduced some soft-phones into the system, though large-scale provision must wait until we have the additional bandwidth which will come with Silk-2. We have provided enhanced facilities for Quality of Service provision – for successful VoIP over congested satellite channels. We have also provided some priority channels for this traffic. We have devoted some attention to security problems – particularly to putting minimal holes in firewalls. Finally, we discuss the further developments planned for the future.

1. Introduction

This Deliverable was supposed to describe the facilities available at the end of the SPONGE project for Voice/IP (VoIP). As such it was supposed to be much the same as the VoIP part of a SPONGE Deliverable of a similar name. When we looked more thoroughly at the problems of deploying VoIP for the OCCASION User community, and at the changes that had been made in the past year in the VoIP services used at DESY and UCL, we decided to adopt a quite different approach. We wanted to adopt something that could be replicated and expanded in the OCCASION community.

In SPONGE we had a limited number of Cisco 7960 IP telephones, and a very limited bandwidth available. We were also running Cisco VoIP servers at both UCL and DESY. For these two reasons we decided to restrict ourselves, in SPONGE, to the Cisco 7960 telephones. Because UCL was committed to a number of developments with SIP and IPv6, UCL wanted to pursue a SIP [2] approach – which could also embrace IPv6. Because they were running production systems, and had no actual project on IPv6, DESY ran a Cisco Call Manager based on the Cisco proprietary protocols called SCCP or Skinny [3]. The Call Manager software available until mid-2006, Release 4 (R4), has only limited support for running Skinny and H.323 together, with no support for SIP. Cisco have an alternate voice server that supports SIP, "Cisco SIP Proxy Server Version 1.2". The software used in DESY had no SIP support configured, while that used at UCL ran only SIP. As a result the two organisations were running incompatible systems. We knew that this would be overcome in the next major Cisco Release (R5), so we decided initially to continue to pursue both directions at the beginning of the OCCASION project. In early 2006, it became clear that the R5 software would not really be usable for OCCASION until late in the year, and the OCCASION partners were quite happy to wait with IPv6 until 2007. We therefore decided that running the two versions would only confuse the OCCASION users, and that we would concentrate on the Skinny versions for the first year – except for the Exco meetings which dial out via UCL into the Cisco audio-conferencing system..

Although the above decision simplified the systems we needed to run, it also reduced drastically the number of soft-phones that could be used. There are a number of open-source and commercial SIP softphones available. However there only two Skinny softphones are available – one from Cisco, and another from IPBlue [4]. The deployed IPBlue phone has a free demonstration release, which requires a license to be purchased for serious usage. DESY is running large-scale VoIP services, so that they cannot easily change releases of their Call Manager software. UCL-CS is doing things on a much smaller scale, and can therefore afford to experiment. During the first year of the project we determined to run an all-Skinny system based on Cisco R4, but to use multiple Call Managers. We also would use both hard-wired and soft-phones. These decisions would allow much larger deployments of VoIP, with local autonomy.

While we are discussing with Cisco the possibility of further donations of equipment, the current trajectory, which is described in this Deliverable, allows extension of VoIP to the OCCASION User community at modest cost.

2. The Cisco-based System

2.1. Overview of system architecture

A voice over IP system typically consists of a number of components: telephone(s), Voice Server(s), MCU(s), PSTN gateway(s), and other components. This is illustrated in Figure 1.

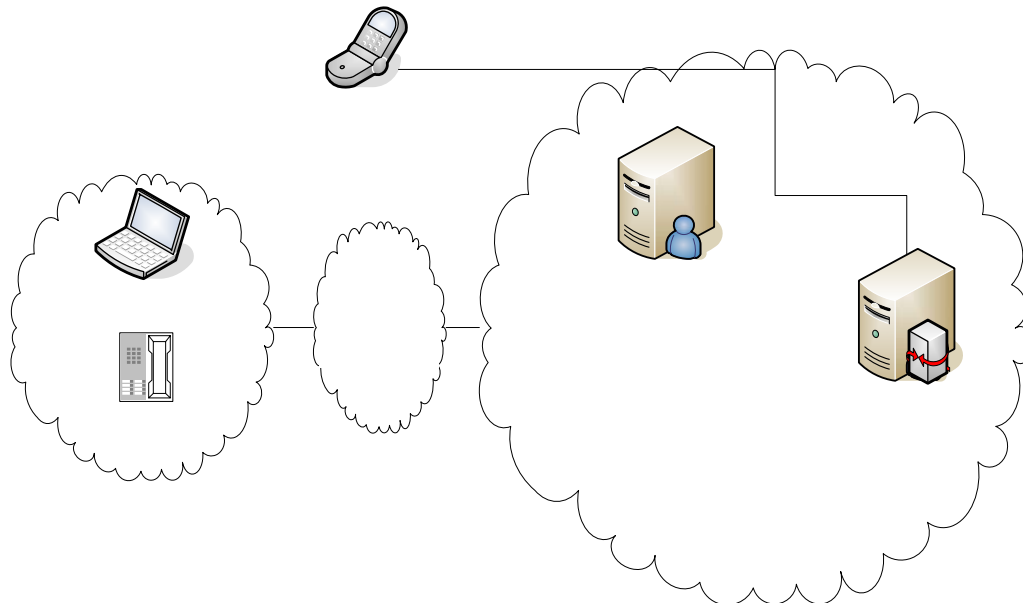


Figure 1 Schematic of VoIP System

Figure 1 Schematic of Skinny Voice/IP Architecture

The telephone can be one of three devices: a normal telephone, an IP phone, or an IP soft-phone. An ordinary telephone is connected to the General Switched Telephone System (GSTN), which may be a private system or the public one. An IP phone has a similar physical interface device, but implements a specific VoIP protocol and is connected to a data network. A soft-phone is just an application, which provides VoIP functionality that runs on a computer with a soundcard, microphone, and loudspeaker.

The second component is a Voice Server (e.g. Cisco Call Manager); this serves all the functions of a PABX. Thus, as with the GSTN, there can be many Voice Servers, which can talk to each other. As with the GSTN, there is a protocol for setting up calls, which performs the signalling between the telephone and the voice server(s). One standard such VoIP protocol is the SIP, which has been standardised by the Internet Engineering Task Force (IETF). Another is the Skinny Call Control Protocol (SCCP), which has been developed by Cisco and deployed at DESY for control of IP telephones. Both protocols allow signalling between servers, and between server and VoIP telephones.

There are further protocols for multimedia conferencing; one is based on H.323; another is based on the Mbone tools. They are of interest to Silk/OCCASION, but will be treated in a separate report.

In addition there exist VoIP architectures, such as Skype, that are self configuring and do not require explicit central management, apart from user authentication. Skype is mentioned here, and has had considerable deployment – even in the Silk countries. It is not adopted as an official OCCASION offering, because the way routing is done MAY result in uncontrolled usage over the Silk network.

2.2. Brief overview of SCCP

The Skinny Call Control Protocol (SCCP) is a Cisco-proprietary protocol, which is used for signalling to their VoIP clients. It provides for more functionality in the Servers with enhanced management features. The basic building blocks are still like those of Fig. 1, with some key differences. Because SCCP is proprietary, there are very few soft-phones available for it so most users deploy the Cisco IP hard-phones. However Cisco provides an SCCP soft-phone, the Cisco Communicator, and another is available from IPblue.com. On the other hand the use of the system from the user's viewpoint has been made as simple as possible. There is

Remote site

IP Blue softphone

IST-026546 D4-1 Voice over IP Facilities under the OCCASION Project

almost no configuration capability by the end user; it is all done by the systems administrators – and heavily automated. Because it is a self-contained system, Cisco has been able to put in many features, which are not in the standardised SIP protocols. In the next version of the Call Manager (R5) Cisco plan to increase the SIP feature set, so that it is more in line with that of SCCP.

The building block that does the actual routing and management of calls is the Call Manager. This is equivalent of the Proxy in Fig. 1. The current version R4 of the Call Manager support SCCP, H.323 and SIP on the trunk lines, but only SCCP to the end telephones. The R5 version of the Call Manager, due for routine service late in 2006, will support also SIP to the end user, using a B2BUA model. There is no plan to support IPv6 in the near future on the Call Manager, because Cisco claims that there is yet no commercial demand.

2.3. Audio Conferencing

The current releases of the Call Manager have a function called “Meet Me”. When that function is enabled, it is possible to set up calls to the particular “Meet Me” extension, and all the telephones so calling are able to speak to all the others and to hear all the others.

3. The Implementation of VoIP for the OCCASION Project

3.1. Introduction

This Deliverable describes the services we have developed for the first eighteen months or so of OCCASION. During this period we wish to be able to support extensive point-point VoIP between people on the Silk NRENs and others on NRENs in Europe and elsewhere. We also wish to support audio-conferences with 10s of participants on equipment attached directly to the NRENs. It is inevitable that academics are fairly mobile, and may be away from their computers for some important audio-conferences; for this reason we want to be able to support some access from ordinary telephones or mobiles. We certainly do not intend to support a full-blown VoIP service using the Public Switched Telephone Networks in Europe. We would like to support both hard-wired and soft-phones. We also would like to have facilities that can be expanded to a significant numbers of terminals – without it becoming an impossible load on the OCCASION management.

We would like to be able to accommodate video when this is possible, though this is the subject of another Deliverable.

3.2. Call Managers

Based on the facilities of Section 2, we have set up two Call Managers (CMs) – one at DESY and one at UCL – using Skinny; both use the CM R4. The DESY CM is attached to their normal PABX; currently most of the Silk Cisco telephones are registered on the DESY CM. The UCL CM is connected to the DESY one, and we have resolved the necessary firewall holes to allow the connection of the phones attached to UCL CM to talk to terminals attached to the DESY one. Firewall and registration security problems are still being encountered, but they are being resolved as they are met.

At DESY we have set up a “Meet Me” extension number, which allows audio conferences of up to 20 participants to be in an audio conference. The same facility will be added shortly at UCL.

This configuration has resolved many of the problems which will arise when the system is extended. Then the other NRENs may decide to put in their own Call Managers – connected to DESY via the Silk Network.

3.3. PSTN Integration

At DESY, we have authorised dial-out from the PABX under password control. Currently each individual extension has its own password and generates its own bill. We are not yet constraining calls to be only local or national; this would be easy to implement if it was needed, because we have all the management facilities we would need on a PABX. Inward calls are possible to any extension – and hence also to the Meet Me number. Any number directly on the DESY exchange can be dialled directly; any number on the UCL exchange requires a prefix.

At UCL we have a PSTN gateway – though currently with only 3 lines and running SIP. We have authorised out-going calls to be made only to three numbers: the Cisco Conference System, the BT speaking clock and Kirstein’s home. This use of the Cisco system allows us to have large conferences with local number dialling

IST-026546 D4-1 Voice over IP Facilities under the OCCASION Project

– wherever Cisco has offices. Again inward calls are possible to the small number of VoIP SIP extension registered on the UCL SIP Server.

3.4. Clients

Currently we support only two clients – the Cisco VoIP 7960 telephone, the IPBlue soft-phone [4] and the Cisco Communicator [5]. We presume that it will be straightforward to add other clients – but this has not been done yet.

3.5. Security

Security problems have caused the biggest headaches so far – and remain such. We will outline some of the problems we have had to overcome. First there was the configuration of the DESY firewall, so that calls from the UCL Call Manager could be accepted at all. This required information from Cisco on how to configure the firewall.

DESY is willing to allow specific terminals (IP numbers) to come in through their firewall. At present they allow only specific static ones; when they receive an upgrade of their stateless packet filter to a dynamic stateful one, dynamic IPs will be permitted. Anyone coming into UCL from DSL links (which Kirstein uses), would normally receive a dynamic IP number – either from the DSL server, or from the VPN that UCL-CS mandates for external connections; the VPN manager has had to be modified to provide static IP numbers – linked to the log-in of the VPN. The Cisco Communicator is configured at DESY so that it has the same extension as the PABX one for the person, and can be used when the person is absent; because of the firewall considerations, it can be used only via a DESY VPN. In the future, when they have upgraded their system, they will allow any voice traffic in as long as it's done via H.323, SCCP and eventually (after R5) SIP.

The client's registration to the Call Manager also has security implications. The UCL mechanism currently allows self-registration; this is an effortless way to bring in new people. The DESY set-up requires prior insertion of the MAC address of any incoming terminal; this is more secure, but requires manual insertion which can be manpower intensive.

So far there is no encryption enabled on the data paths; this could be added. We must study whether this is straightforward with CM R4.

3.6. Quality of Service

We have enabled bandwidth priority on the routers at the ingress and egress to the Silk network. This provides adequate QoS for the limited number of calls so far. We have provided a total of 1 Mbps for real-time traffic. The checking of the authorisation has proved a serious load at the DESY hub of the VSAT system. Cisco has eased that problem by donating a GSR router to enforce the flow priority; this has made that problem disappear.

4. Extensions Enabled for OCCASION Purposes

Of course all members of DESY are able to use the DESY system. This section will discuss only those telephones which have been set up so far for OCCASION purposes. Here there are three columns for the extensions:

- Those registered with the DESY Call Manager
- Those registered with the UCL Call Manager (UCL-CM)
- Those registered with the UCL SIP Server (UCL-SIP)

The UCL SIP Server is used at present for the weekly Exco meetings, because of the way it is connected to the Cisco internal audio-conferencing system; only IP phones with their clients set up for SIP can use the UCL SIP Server. The UCL-CM is currently being used mainly for testing

IST-026546 D4-1 Voice over IP Facilities under the OCCASION Project

	DESY-CM	UCL-CM	UCL-SIP	Comments
Western People				
Andrews		2900		
Berezhnev				No SIP phone
Frese	2588			Also uses number with Cisco Communicator
Janz		2913		IPblue-office
Kars	3313			
Kirstein		2902	2011	Office
Kirstein		2914	3004	IPblue-office – demo; SIP phone is via PSTN
Kirstein - Home		2912		IPblue-home - demo
Kirstein-mobile		2906		IPblue-mobile
Lavrov	4085		2810	SIP Phone Not Configured
NOC	4466			
O'Hanlon		2905	2012	CM Phone is IPblue-office
Socrates		2903		IPAQ- IPblue-office
Meet-Me	5939			
NRENs				
AF	5922			SIP Phone Not Configured
AM1	5929			
AM2	5928			
AZ	5926		2511	
GE	5924		2111	
KG	5920		2310	SIP Phone not registered this year
KZ	5927		2611	
TJ	5921			SIP Phone Not Configured
TM	5923			SIP Phone Not Configured
UZ	5925		2710	
Other Resources				
UK Speaking Clock			3002	Clock 123
Cisco TC			3003	x+44-20-8824-8777
Main Switchboards				
DESY				X+49-40-8998-xxxx, dial directly with ext
UCL-SIP				x+44-20-7387-8781, 2nd dial-tone +SIP ext

Figure 2. The Extensions Used at Present

As currently configured in the CMs, any local extension can be dialled through the DESY CM with a 4-figure number. By contrast, any UCL extension dialled via the DESY CM must be prefixed with 6353, while any DESY extension dialled through the UCL CM must be prefixed with a “1”. On the other hand, the number showing up on the telephone as the distant one, is just the four-figure extension. This anomaly is expected to change shortly.

5. Experience So Far

The implementation of the facilities described so far has taken much of the effort available in this area. We did not want to open up the system before we had at least two CMs talking.

Up to June, we were using only the Cisco IP 7960 phones registered on the DESY CM for mainly person-

IST-026546 D4-1 Voice over IP Facilities under the OCCASION Project

person administrative calls. Once we moved to have two independent CMs, we introduced also the IPBlue software client. This is now a regular part of our services – though only a very small number are deployed. The demo version of IPBlue is free, and has the same functionality as the purchased one – except that it needs restarting every two minutes, with the session restarting every 20 minutes. The demo version is very useful for all testing purposes, though the need for restarting makes it a nuisance for proper conferences.

We have used the previous Cisco audio-conferencing service for regular Silk Exco purposes. For this the limited number of Silk participants this has worked well. For full OCCASION audio conferences, this would be very limiting – and would involve Cisco unnecessarily.

We have used the system for an increasing number of point-point discussions – though the Cisco phones are still sited mainly at the technicians' desks. We have had up to ten people at a time in the Meet Me audio conferences – with four coming in from Silk NRENs. We are currently holding these on a weekly basis, to shake down the system by the 4th quarter of OCCASION.

6. Regulatory Considerations

The DESY Acceptable Use Policy must permit only traffic that is permissible by the DFN – the German NREN. There are some strong regulations that have been put in place by the German Regulatory Organisation. They are based on more general European Commission directives. The German regulations are described in: [6] and [7]. It is not our intention to describe the technical requirements, legal constraints and penalties described in these regulations. Suffice to say that the technical requirements are so serious, that neither DFN nor DESY are prepared to allow any traffic that could make them liable to be responsible under these laws.

There are certain cases where the operator is not obliged to provide the technical information requested; these are:

- If it is a closed group of less than 1000 subscribers
- It is only for Internet traffic, with access to the Open network (i.e. telephone network).

While neither of these applies to DFN itself, they may seem to apply to the Silk uses. Another part of the regulations states that if a network is purely a transit network, then it does not have to abide by the need to keep the records.

In view of the above, VoIP for Silk/OCCASION should obey the following constraints.

1. Potential VoIP users from the Silk NRENs must have their IP numbers reported to DESY for incorporation in a DESY database. This is required both for the regulations and to ensure adequate QoS.
2. DESY may regulate the number of simultaneous users permitted from each NREN.
3. The users in the Silk countries, who have access to the telephone network via DESY should be listed explicitly by message to the Exco.
4. The persons whom they should be permitted to communicate with via the telephone network should be listed explicitly by message to the Exco.
5. Any telephone charges that will be levied by DESY should be agreed between the member of the Silk Board for the NREN and the DESY member of the Silk Board.
6. A mechanism for collecting the charges in (5) must be agreed that is satisfactory to the DESY member of the Silk Board.
7. The constraints on usage by devices in the country of a specific NREN must be in accordance with the National regulations of the country concerned.
8. The Silk Exco may impose additional constraints at any time, if these are requested by parties with a legitimate reason to request them.

7. Further Developments Planned

We will concentrate next to bringing our H.323 video-conferencing facilities to the same status as the VoIP is now. This has a high priority, in view of the 130 K€ investment that has gone into the provision of the hardware.

For the VoIP, our first priority is to ensure that all the Silk partners participate in the audio-conferences with their existing Cisco 7960s. We will then encourage some sites to deploy more soft-phones – particularly for their users. This will require some implementation of QoS on the NRENs. We will not try to have a large expansion of soft-phones until we have more bandwidth with Silk-2 – which is not expected before the first quarter of 2007.

Around that time, we will also plan to introduce R5. This will allow far better integration of the video and the audio services. It will become easier to integrate the VoIP and H.323 services – using also SIP. We hope at this point to be able to introduce IPv6, and benefit from a substantial donation from Cisco. By the end of the project we would like to see VoIP in regular use by the users.

References

[1] Resources for H.323 and VoIP in the SILK Environment, Deliverable 13, the SPONGE Project,

<http://www.silkproject.org/sponge/SPONGE%20documents/SPONGE-D13%20Resources%20H232%20and%20VoIP.pdf>

[2] M. Handley, et al, *SIP: Session Initiation Protocol*, RFC 3261, June 2002

[3] Cisco SCCP: Skinny Client Control Protocol, <http://www.javvin.com/protocolSCCP.html>

[4] IPBlue Free Downloads, <http://www.freedownloadcenter.com/Authors/IPBlue.html>

[5] Cisco Communicator,

http://www.superwarehouse.com/Cisco_IP_Communicator/SW-IPCOMM-E1-CH1/p/387591

[6] Telekommunikationsgesetz (TKG §110),

http://www.gesetze-im-internet.de/tkg_2004/index.html

[7] Telekommunikationsüberwachungsverordnung (TKÜV),

http://www.gesetze-im-internet.de/tk_v_2005/index.html.