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SILK Project Operations Networking and GEANT Extension
SPONGE
Deliverable D4: Voice Over IP in the Silk Environment
- Resources required, Parameters needed, and Experiences

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Editor(s):	**Peter T. Kirstein**
Participant(s):	**P. Kirstein, J. Andrews**
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Abstract: The Deliverable specifies the set-up provided for Voice/IP in the Silk Project. It describes the configuration set up, the equipment in the remote sites and in West Europe, the mode of usage, and practical experience.

Keywords: Silk Project, NATO Networking Panel, Southern Caucasus, Central Asia, Satellite Internet Access, Voice/IP, Audio Conferencing

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

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1 Introduction

The Silk Network is operating in countries where salaries are low compared to the West, and the cost of communications is high. Nevertheless, it is very important to ensure that it is possible to have good communications between the participants in the project. Moreover, the existence of the Silk Network makes it possible to install both Voice/IP (VoIP) and Multimedia/IP (MMIP) facilities. This Deliverable is concerned with the VoIP services.

In the context of the SPONGE project, we decided first to concentrate on exploring the services with the SPONGE partners – under the SPONGE WP 4, and then to consider what to do in the wider Silk context. With the assistance of Cisco, we put in both IP telephones and an audio server. For some purposes we put in also Public Switched Telephone (PSTN) lines and arranged for them to dial into an audio conferencing system run by Cisco. This report describes, in Section 2, the system set up. Initially the performance was very poor for audio; we describe in Section 3 what had to be done to improve performance. After we had shown that the system could provide the facilities we wanted, we requested additional equipment from Cisco; the current status of the work is described in Section 4. Finally, in Section 5, we describe what further work we plan to do.

2 The System Installed

The system installed at each remote site is shown symbolically in Fig. 1. At each site there is an NREN, with one or more IP telephones (IPT) connected to an Ethernet (Shown in Fig. 2, where for Internet read Silk Network, and the PSTN exists only at UCL and DESY. This Ethernet net is connected, through the NREN's local or National networks to the Cisco Router (CR) of the relevant country. At UCL we have an Audio Server (AS). AS has two functions, it serves as a switch, and also registers the remote users. Each remote telephone is assigned a particular four digit telephone number. The act of dialling a four figure number transports an IP signalling packet to the AS, which connects to another telephone representing the number called. The AS then dials the telephone corresponding to that number just the same as a normal telephone switch. Several signalling protocols have been for this purpose. The AS supports both a proprietary Cisco signalling format, and the IETF-standardised Session Initiation Protocol (SIP). We have chosen to use SIP.

These Cisco telephones are able to connect to three remote telephones at the same time, so a four-way telephone conversation is possible. It would be possible to connect in many more telephones to the same call, but this would need additional equipment which does not currently exist in UCL.

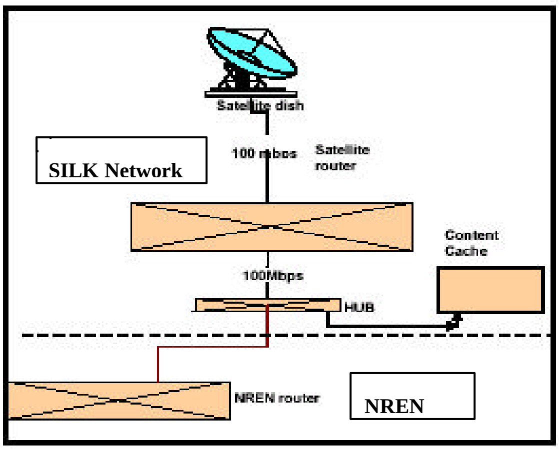

Figure 1 Schematic of Earth Station and Silk Router configuration

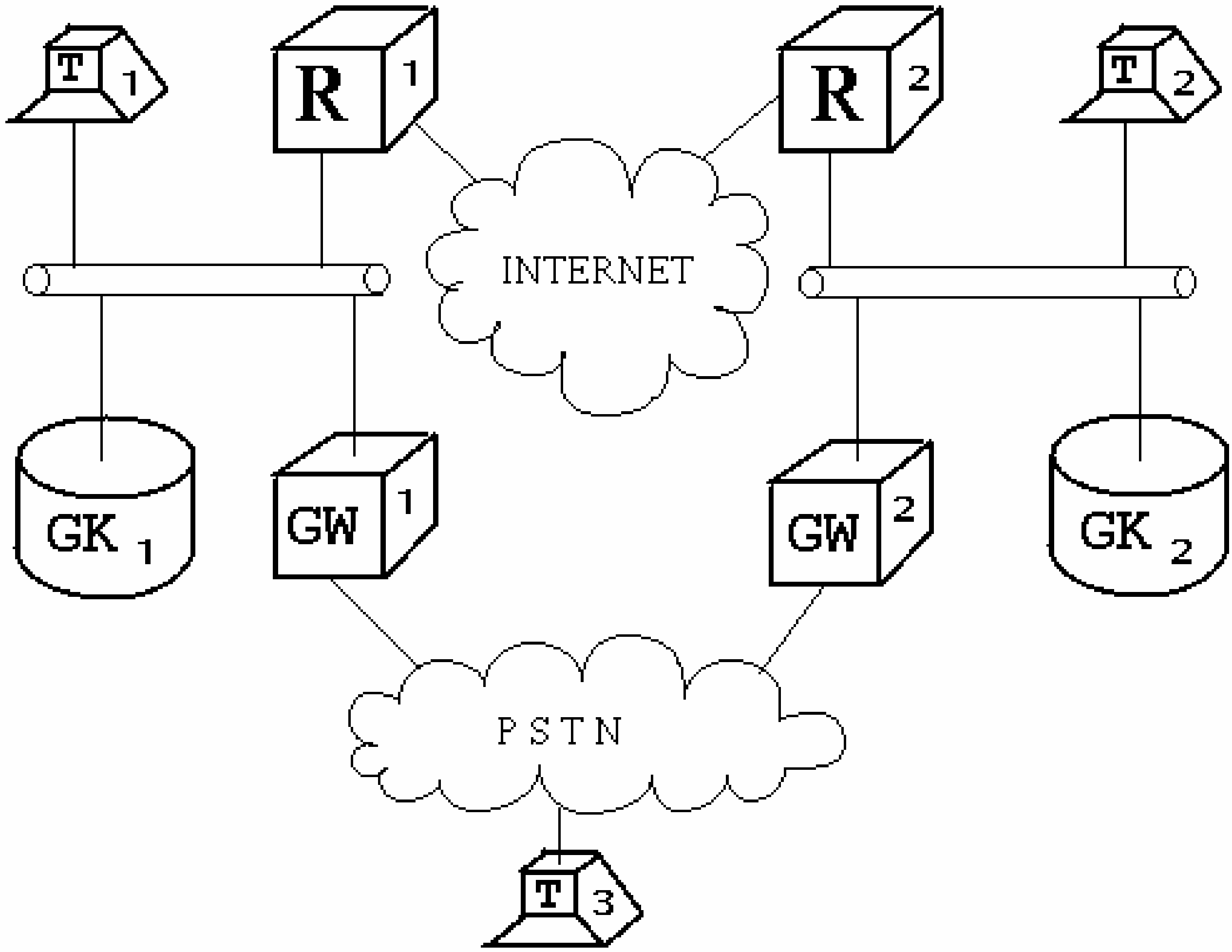

Figure 2 Schematic of normal teleconferencing configuration with a Gatekeeper at each site

If one would like to provide Quality of Service, as is needed in video conferencing or large-scale audio, one puts a so-called gate-keeper, shown as GK in Figure 2, at each site. If there is less priority traffic, as is currently the case in the Silk audio work, then the mechanisms of Section 3 are adequate. In the current work no gatekeepers are configured.

The AS is connected to the Public Switched Telephone network (PSTN) – currently with up to 3 lines; it would be quite possible to add more. It would be possible to configure the Server so that the remote telephones could dial an arbitrary telephone number. This feature has been deliberately omitted; only a few important external PSTN telephone numbers have been programmed in – again with a four-figure number.

Telephone i at site AA is given the number Aai; i.e. the five telephones at site 10 might be given the telephone numbers 1001, 1002, 1003, 1004 and 1005; those at site 21 might be 2101, 2102, 2103, 2104, 2105. The system is currently configured so that if remote 1002 dialled the number 2103, this number would be passed to the server AS at UCL. AS would then connect to the telephone 2103 via a SIP call through the Internet, and the call would be established. It would be possible for 1002 to dial both 2103 and 2304; then a three-way conference would be established.

We have registered several PSTN numbers on AS; for example, the number 02076797286 might be assigned the number 9001. If 1002 dials 9001, then 9001 is passed to AS. AS will then dial the number 02076797286 through the PSTN, and a call between 2001 and that telephone will be established.

Cisco runs a large audio-conferencing system, with local or national number access in almost every city, world-wide, where it has offices. Audio-conferences can be set up by dialling one of these access numbers, and then dialling several more numbers including that of the unique teleconference that has been established; dozens of simultaneous users can be supported in each teleconfernce. Knowing that a specific teleconference is occurring at a particular time serves as an elementary form of access control. Three of the

UCL PSTN ports on the AS have been programmed to dial in to the Cisco audio-conferencing system. Thus three remote users could connect in to an audio conference together with many more users coming into the Cisco system directly through the PSTN.

While it is particularly convenient to use the IP telephones, it is also possible to use an ordinary laptop or desktop PC with the relevant sound cards and software. These are called Soft-phones, and must support both SIP and audio. Netmeeting is one such piece of software, but there are many others such as Bonephone and K-phone.

3 Quality of Services Enhancements

Depending on the audio coding used, and the quality of audio desired, it takes between 2.4K and 400 Kbps to support a telephone call. Normal telephones will normally be run at 64 Kbps, but a GSM cellular phone normally runs at 9.6 Kbps. Currently many of the remote sites only have access to the Silk network at around 256 Kbps for the whole country – though this is expected to go up to around 500Kbps from NATO sources. As a result the satellite channel is often very congested during the working day. Normal data transfer uses a reliable end-end transport protocol like TCP; this ensures that lost parts of the data stream are re-transmitted to ensure reliable transmission. Any such mechanism causes very significant delays during transmission over busy channels. Any delay over 250 ms starts to get annoying to the user, and even a single retransmission request over a satellite would require more than a 750 ms delay. Hence for audio, reliable transport is not used, and after a certain delay, the packets are dropped at intermediate nodes. With some coding schemes, e.g. RAT [], there are error-correcting mechanisms that can partially recreate a missing talk spurt. Such mechanisms are not used in the Cisco IP phones. The quality of the audio becomes most unpleasant with a loss rate exceeding 5% of the packets – which can easily occur in practice.

In the Silk Network, congestion can occur both in the National distribution network and in the access to the Silk satellite channels. In practice, all the IP telephones in the Silk countries are normally located close to the Silk earth-station – usually separated from it only by an Ethernet. Under such conditions, the main losses occur at the access to the Silk earth-stations. In order to ensure that VoIP of reasonable quality can be provided, we have enabled QoS queuing in the Silk Cisco routing. The mechanism is illustrated in Fig. 3.

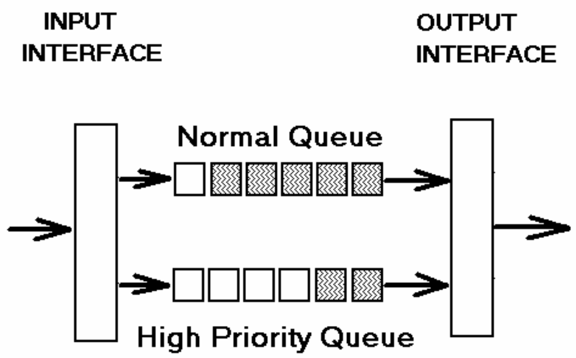

Figure 3 Quality of Service Queuing

It is possible to instruct the Cisco router to assign a high priority to the packets from a particular IP device – i.e. a Cisco telephone. It would probably be possible to apply the same configurations with a soft-phone, but we have not yet investigate the relevant configuration details required.

Using this mechanism, it is possible to carry out audio-conferencing at any time; no special set-up is needed. Of course the audio would lead to a slowing down of all the other traffic going through that interface, but that is not really perceptible with only one audio stream operational. It would, however, be much more serious if there were many such streams.

UCL is working with Cisco using another soft-phone variant and the VOCAL system to provide not only QoS but also admission control – to limit the number of calls that can be started. This work is under the auspices of the 6NET project, not SPONGE. We may try to introduce this into SPONGE at a later stage, since it is obviously relevant.

4 The current status of the Installation

The first use of VoIP was, as was stated in the SPONGE Description of Work [], to demonstrate the technology amongst the SPONGE partners; only later would there be a wider deployment to the Silk partners. Indeed, the Cisco IP phones were given to the SPONGE partners in Armenia and Georgia at the beginning of the project – during the Zagreb Silk Board meeting in September 2002. We went a little further; since we intended to use the units for Silk Executive Committee (ExCo) meetings, we provided them also to our Kyrgyz Republic partner – who is also on the ExCo. At that time, we configured and registered the phones on our Server before shipment. We also configured the Server, installed three PSTN lines, and configured the dial-out both to the Cisco audio-conferencing local access number in London and one home number – to allow conferencing at unsociable hours.

That system was not satisfactory until the QoS modification of Section 3 was made; since that time the quality of the audio has been normally very good. Georgia has the most highly developed of the Silk sites, and were the most frequent user of the system. The Armenian experience has been poorer; this is not due to the audio quality, but to non-technical factors. For a while we used the normal Internet for the audio; suddenly communication failed, because the authorities decided to block the VoIP port used by the Service Provider. Moreover, the regulations are very stringent in Armenia on where equipment may be located; it must be at a site belonging to the NREN – not to its constituent universities. The Kyrgyz usage was of good quality until recently; then the Silk transmitter failed, and has not yet been replaced. They continue to access the Internet by another route (with Silk providing the return path), but that route could not support the VoIP.

The technical quality was so good, that we decided to include to VoIP units with the Standard equipment from March onward. The equipment was so useful, that the Silk Network Operating Centre in DESY configured some of the later telephones to point to a registry in DESY (unbeknown to the SPONGE team!). This meant there could be no communication between the two sets of phones. As soon as the situation was realised, it was agreed to register all the phones on the UCL registry – including those at Groningen (another SPONGE partner) and DESY. We are currently in the process of the new registration; the instructions to achieve this are given in Annex A.

The status of the current installations is given in Annex B.

5 Work to be Done

It is important first to ensure that all the IP phones – both hard and soft – are registered on the UCL Server. Then we must configure the system to work with registration on multiple servers – by communication between the Servers. The Admission Control from the 6NET work should be transferred to the SPONGE/Silk environment.

We will then concentrate on the multimedia activity – using both H.323 and Mbone.

Annex A Details for SILK sites to connect Cisco IP telephones to the UCL SIP server

Each phone needs a global IP address - not a NAT or private address, and should not be blocked by any firewall for UDP traffic. The IP address and the MAC address needs to be registered on the UCL server as the first step.

Please mail J.Andrews@cs.ucl.ac.uk with the MAC address of each phone (this is printed on the bottom of the phone near to the network port and is 10 hex digits usually starting 00), and the IP address that will be used for the phone. If the phone has an external power supply, please power up the phone with no network cable connected. This may take a few minutes and the green then red lamps should light followed by the cisco logo on the display. When the display shows "ethernet disconnected", press the settings button (the bottom right button just below the middle button in the group on the right side). Move down (with the down key) to "status" and press select. Move down to "Firmware" and press select. Note the application and boot load IDs and include them in the email.

If there is more than one phone, please repeat this for the others.

When it is confirmed in an email reply that the phone is registered with the UCL server please configure each phone separately so that one phone is fully working before the next is setup. The phone will not work unless it is registered with the UCL server.

To configure the phone please power it up with the network cable disconnected (if the phone has a power supply - some phones are powered from the network cable so this does not apply).

When the display shows "ethernet disconnected" (or "configuring VLAN) press the settings button and move to "network config" and press select. Press * * # to unlock the phone. Move down to "Erase config" and press yes, then save, then save again.

Press settings and move to "network config" and press select

Move down to "DHCP enabled" and press no. Move up to IP address and press edit. Enter the IP address for the phone (use the number keys and the . (dot) key; use < to correct if needed). Press validate. Do the same for the "network mask" and "default router 1".

Enter the TFTP server as 128.16.67.1 and validate. Press save and save again. Power reset the phone and connect the network cable.

The phone should show the green the red lights, then the display should show the cisco logo and then "configuring VLAN", then "configuring IP". It should then start loading the SIP code. This may take several minutes and show the red or green lights. The phone should then automatically restart and repeat the loading again taking several minutes. Once complete, the phone should display a SIP phone number and be ready to use. There should be a dial tone and calling 3002 should connect to the UK speaking clock as a test.

(The SIP code loading only happens the first time the phone is used. The code is loaded into the phone's memory so when power reset should restart quite quickly.)

Annex B Current Assignment of Telephones for SPONGE/Silk

Number	Registered User	IP Address
2010	Andrews, J	128.16.67.11
2011	Kirstein, P	128.16.67.10
2012	O’Hanlon, P	128.16.67.12 (also used for Vocal)
2110	Ramaz,GE	217.147.227.62
2111	Ramaz,GE	217.147.227.61
2210	Ruben,AM	217.113.0.58
2211	Ruben,AM	217.113.0.59
2310	Askar,KG	212.122.96.197
2311	Askar,KG	212.112.96.196
2410	Janz,R	129.125.3.77/129.125.44.197 (softphone)
2413	Aleksander,PL	149.156.97.144 (softphone)
3002	Speaking Clock	(123)
3003	Cisco Audio Conf	(020-88248777)
3004	Kirstein, P Home	(83403154)
3005		
3006	BT Audioconferencing	
3006	BT Audioconferencing system	

There are another 10 telephones in the Silk sites, which have not yet been connected in to the UCL server.

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2.