

B Appendix IP Telephony Hardware/Software

> **Product Name: Windows Messenger**

Product URL: <http://www.microsoft.com/windows/messenger/>

Vendor: Microsoft

Supported Protocols: SIP

Platform: Windows

Description: A well-know SIP softphone from Microsoft. There are different versions of the software and not all of them support SIP. This is unfortunately confusing. The most standard -compliant seems to be version 4.7. The phone is easy to use, but there are several violations of specifications. It supports audio, video, instant messaging and white board.

The client sends **BYE** instead of **CANCEL**. Microsoft uses a 'proprietary extensions of the presence' document and some other software vendors implemented the extensions as well. The realm of digest credentials must be same as the hostname of the server; otherwise the client refuses to authenticate.

New versions of Windows Messenger use **INVITE** to establish a session before an instant message is sent and thus are incompatible with other user agents, even with older versions of Windows Messenger. It does not support any NAT traversal techniques but it can be used from behind NATs with some aid from a SIP server in the public Internet because it implements symmetric media and signalling.

> **Product Name: kphone**

Product URL: <http://www.wirlab.net/kphone/>

Vendor: Wirlab

Supported Protocols: SIP

Platform: UNIX/Linux

Description: A user agent for Linux. Versions up to 4.0 were linked with KDE, starting from 4.0 kphone. It does not require KDE libraries anymore and uses qt only. The phone supports presence and instant messaging based on SIMPLE standards. Video conversation is supported using vic. The phone is not very stable yet and crashes from time to time. It supports G.711, GSM, and iLBC codecs.

> **Product Name: Linphone**

Product URL: <http://www.linphone.org/>

Vendor: Simon Morlat

Supported Protocols: SIP

Platform: UNIX/Linux

Description: It is a simple SIP user agent for Linux and uses the GNOME frameworks. The phone is unstable and crashes from time to time.

> Product Name: X-Pro, X-LiteProduct URL: <http://www.xten.com/>

Vendor: Xten Networks

Supported Protocols: SIP

Platform: N/A

Description: It is a good softphone with many interesting features. It has sophisticated NAT detection mechanisms. SIP support is good. There seem to be no interoperability issues (as far as we have tested). X-lite is a free version; X-Pro must be purchased. The softphone is also available also for WindowsCE-based PDAs.

B. 2 Hardphones**> Product Name: Cisco 7960**Product URL: http://www.cisco.com/warp/public/cc/pd/tlhw/prodlit/7960_ds.htm

Vendor: Cisco Systems

Supported Protocols: H.323, SIP

Platform: N/A

Description: Operational Experience: It supports H.323 under the assumption of having a Call Manager, which is a Cisco software product to manage Cisco hardphones. The communication between Call Manager and Cisco 7960 is done using the Cisco proprietary Skinny protocol (not a standard H.323 protocol!) and therefore the standard H.323 communication with an external client is only performed with the Call Manager and not with the phone itself (limited standard compliance of the phone). It supports also SIP protocol but from a standard point of view. With the following firmware (Application LoadID: POS30203; BootLoadID: PC03A300. It was found to be perfectly interoperable with VOCAL system and with MSN messenger. Please pay attention that Power Cord is not included in default selling configuration. Take care when buying it or delays will occur before you can effectively use it (because of Cisco vendor's slowness). Overall Evaluation: Good hardphone. It is a little bit too expensive and not completely H.323 compliant. The SIP part is very good and standard-compliant.

The phone has excellent design. It is very comfortable to use, even during long conversations. The display is big enough to be seen even from a great distance. The phone has six lines. A SIP image has to be loaded into the phone before it can be used as a SIP phone. SIP-standard compliance is very good up to 6 SIP accounts can be registered simultaneously.

Unfortunately it is not possible to decline an incoming call. The phone can be switched into Do-not-disturb mode, though. Short packets containing just four zeroes (often used to keep the NAT bindings alive) freeze the phone. This should hopefully be fixed in the latest SIP model (not tested yet).

> Product Name: Adtech SI-150 IP PhoneProduct URL: <http://www.adtech.be/text/si150.php>

Vendor: Adtech

Supported Protocols: H.323, SIP

Platform: N/A

Description: Operational Experience: It supports H.323 but with a particular version of the

firmware we have experienced some interoperability problems with standard H.323 clients (NetMeeting). Regarding the SIP part, with the firmware version SIP 2.07.06 CS 49AC, it showed some interoperability problems with VOCAL systems and MSN messenger. Some bug fixes are needed in order to avoid hook and crash problems of called clients. Overall Evaluation: Not yet a mature hardphone (at least with the firmware version we tested). It is quite inexpensive.

> **Name: Komodo Fone (now Cisco ATA)**

Product URL: <http://www.cisco.com/warp/public/cc/pd/as/180/186/index.shtml>

Vendor: Cisco (formerly Komodo)

Supported Protocols: SIP, H.323

Platform: N/A

Description: Black-phone-2-Ethernet/analogue-line adapter, both H.323 and SIP (currently SIP only for the adapter). The adapter supports two lines, only one of them at a time can use G.729 codec; the other one can use G.711 only. It supports symmetric signalling and media and can work from behind NATs. Standard-compliance is good.

One of the nice features is the possibility to create dialling plans using regular expressions. Any analogue phone that supports DTMF dialling can be connected to the adapter.

> **Product Name: BudgetTone-100**

Product URL: <http://www.grandstream.com>

Vendor: Grandstream

Supported Protocols: SIP

Platform: N/A

Description: This is the cheapest SIP phone available at the time of completing this document. The price is very low, but you get what you paid for. The phone is mostly standard-compliant; from time to time there are some interoperability problems but the manufacturer fixes them quickly. The biggest problem of the phone seems to be its HW. The phone can dial numbers only; it supports STUN and symmetric media and signalling. Many codecs are supported. iLBC support has been announced. If you need the cheapest phone available then BudgetTone is your choice. Be prepared to receive a phone that looks like a toy with a not very friendly user interface.

> **Product Name: 5055 SIP Phone**

Product URL:

http://www.netergymicro.com/products/reference_designs/ip_t2.html

Vendor: Mitel Networks

Supported Protocols: SIP

Platform: N/A

Description: The 5055 is a high-quality SIP phone that meets the price/performance needs of business users. Also available is the 5305 desktop conference unit and the 5310 boardroom conference unit. Mitel is a traditional manufacturer of telephones. 5055 has very solid design and is comfortable to use. The phone has many programmable buttons. The display is very small but functional. SIP implementation is good without any serious interoperability problems. The phone is rather expensive compared to other SIP phones available on the market. Audio quality is very good.

> **Product Name: XPressa**

Product URL: <http://www.pingtel.com/products.jsp>

Vendor: Pingtel

Supported Protocols: SIP

Platform: N/A

Description: A hardphone with unusual design. We had some problems with record routing (loose routing) which we were unable to solve so we did not use the phone much.

> **Product URL:**

Vendor: Siemens

Supported Protocols: H.323

Platform: N/A

Description: The former HiNet LP 5100 (in 1999) comes without an integrated switch and supports only 10 MBit/s. The supported codecs include G.711 and G.723. While in theory multiple firmwares can be used on this phone, the most popular was the H.323 (V2) firmware.

Depending on the firmware/application version the phone can be remotely configured via HTTP (using port 80 or 8080 - depending on firmware) and (sometimes only) using a special Windows Deployment Tool that allows the configuration of multiple ones at once.

To upload a new firmware, older versions required the use of DHCP and TFTP to update the firmware and FTP to access the application file.

New firmware versions have been seen booting the phones once in a while - regularly, when the phones aren't registered to a gatekeeper and without obvious reason while registered.

The telephones support the H.323 feature of Registration KeepAlive, but ignore the registration timeouts from the gatekeeper and resend their registrations every 60 seconds. If they are not currently registered the interval decreases to 30 seconds. The telephone does not support the FastStart protocol feature (at least not in the old versions).

> **Product Name: optiPoint 300 basic**

Product URL:

Vendor: Siemens

Supported Protocols: H.323

Platform: N/A

Description: It is a H.323 phone with a limited display (numbers only), less function keys than its big brother 300 had and without G.723 support. It supports H.323 FastStart procedures (10 MBit/s only).

> **Product Name: optiPoint 400 standard**

Product URL:

http://www.siemens.com/index.jsp?sdc_rh=&sdc_flags=3&sdc_sectionid=2&sdc_secnavid=110&sdc_3dnvlstid=&sdc_sid=10240442142&sdc_countryid=0&sdc_mpid=0&sdc_unitid=0&sdc_conttype=3&sdc_contentid=1004068&sdc_ggid=17&sdc_langid=1&sdc_m4r=

Vendor: Siemens

Supported Protocols: H.323, SIP

Platform: N/A

Description: This is a phone with integrated 10/100 Mbit/s mini switch, display and numerous features. Like the predecessor optiPoint 300 advanced, this phone can run different firmwares: HFA (CorNet), H.323 and SIP. The H.323 protocol behaviour and the available features are pretty much the same as for the predecessor (see above).

> Product Name: Snom 100

Product URL: http://www.snom.com/snom100_en.php

Vendor: Snom.de

Supported Protocols: SIP, H.323

Platform: N/A

Description: The phone is easy to use with Graphical LCD display and four Softkeys. Caller Id, Hold, Divert and Transfer, Call Waiting Indication, Message Waiting Indication, Speed Dial, Phone Book, Call and Deny List, HTTP Server, Echo cancellation. The firmware can be upgraded over http.

> Product Name: SoundStation IP 5000

Product URL:

http://www.polycom.com/products_services/0,1443,pw-182-3073,00.html

Vendor: Polycom

Supported Protocols: H.323 (SIP and Skinny announced)

Platform: N/A

Description: This is a conference phone supporting multiple signalling protocols (depending on firmware). The H.323 firmware supports FastStart and Tunnelling (since version 2.5) but up to now (2.8) no H.450. It has very good audio quality and supports Power-over-LAN.

> Product Name: IP Phone 7905

Product URL:

Vendor: Cisco

Supported Protocols: H.323

Platform: N/A

Description: Very simple H.323 phone without H.450 support or integrated switch. It exists in two versions: Either with AC adapter or only with Power-over-LAN support.

This is a cheap brother of popular 7960. The phone has smaller display but the design is as comfortable as in 7960. The SIP image is quite simple but standard compliant. The phone allows registration of just one line, it can dial numbers only and it is not possible to dial a domain (@iptel.org for example). It can be configured through a Web interface but the Web interface is very basic and has some shortcomings.

If you are looking for a cheap but well designed phone and do not mind that you will be not able to dial SIP URIs, 7905 is a good choice. By default it is not shipped with a power supply (Cisco assumes power over ethernet) so do not forget to order one.

> Product Name: IP 200

Product URL:

Vendor: innovaphone

Supported Protocols: H.323

Platform: N/A

Description: Excellent, full-featured (H.450, ...) H.323 phone with a fair sized display. Comes with an alpha-numerical keyboard and eases entering URL addresses. It can access phone-books via LDAP (not tested). Supports overlap dialling.

Early protocol firmwares had a problem when an incoming call has been cancelled by the calling party (receiving a RELEASE COMPLETE while still ringing) when the phone continued ringing and showing a TRAP/error dump on the display when going off-hook. (This problem might be long fixed).

Early hardware had a problem of an incorrectly applied capacitor leading to a fading display (contrast). Contact Innovaphone for a replacement.

> Product Name: i2eye DVD-1000

Product URL: <http://www.d-link.com/products/?pid=8>

Vendor: D-Link

Supported Protocols: H.323

Platform: N/A

Description: A broadband videophone hardware with built-in camera. Video and audio signal can be accessed via cinch connectors to connect television hardware. An external microphone can be connected.

The DVC-1000 has a very minimal H.323 support. H.323 is used to setup a call, while address resolution is done using D-Links LDAP server. The address of the LDAP server is fixed and can not be configured. The D-Link LDAP server can be used for free for all DVC users but doesn't offer resolving other addresses.

To make really good use of the DVC hardware, one needs a proxy instance for dialling that calls the target and the DVC (set to auto reply) at the same time and passing messages through.

> Product Name: VCON Escort

Product URL:

http://www.vcon.com/solutions/videoconferencing/desktop/Escort_Cruiser/index.shtml

Vendor: VCON

Supported Protocols: H.323

Platform: Windows

Description: The VCON Escort hardware H.323 and H.320 client is a PCI-based card that can be installed on any standard PC running a Windows OS. It allows connections at up to 1.5Mbps and includes features for data collaboration (T.120), quality of service (QoS) and interactive multi-cast for allowing viewers to watch a conference over a multicast network. It generally performs well, but compatibility with PC video cards seems to be crucial for trouble-free operation, as in some set-ups, sudden crashes during long operation are not uncommon.

> Product Name: VCON Falcon

Product URL: <http://www.vcon.com/solutions/videoconferencing/group/Falcon/index.shtml>

Vendor: VCON

Supported Protocols: H.323

Platform: N/A

Description: The VCON Falcon is a set-top-box H.323 and H.320 client. It includes a quality camera and microphone. It has a wide array of audio and video connectors to allow it to inter-operate with projectors, screens, multiple video and audio sources, as most often found in a group videoconferencing settings. It is a reliable device, but the remote controlled management interface is difficult to work with and a bit limited (e.g. H.323 aliases cannot include special chars at all).

B. 3 Servers

> Product Name: OpenH323 Gatekeeper - GnuGK

Product URL: <http://www.gnugk.org/>

Vendor: Open Source

Supported Protocols: H.323

Platform: Any platform where you can compile the OpenH323 Library (Linux, Windows, FreeBSD, Solaris, etc.)

Description: Operational Experience: There are some minor problem with Netmeeting. Netmeeting does not support Gatekeeper Discovery, thus you should directly configure the gatekeeper address in the Advanced Calling Options. Netmeeting requests an incorrect bandwidth; disable bandwidth management to avoid problems with GnuGK. It has Radius, MySQL and LDAP support. The manual is written in English, Chinese and Portuguese. It is used in many commercial applications and it has nice graphical interfaces to configure it, to monitor the registrations, to define groups of endpoints, to do call management, etc. Overall Evaluation: it is simply the best Open Source gatekeeper available nowadays. For technical support there is the Gatekeeper Users mailing list.

> Product Name: VOCAL (Vovida Open Communication Application Library)

Product URL: <http://www.vovida.org/>

Vendor: Open Source

Supported Protocols: H.323, SIP, MGCP

Platform: Refer to http://www.vovida.org/applications/downloads/vocal/platform/1_5_0.html for an updated list (Linux, etc.)

Description: Operational Experience: The VOCAL software includes a Session Initiation Protocol (SIP)-based Redirect Server (RS), Feature Server (FS), Provisioning Server (PS), Marshal Server (MS) and Voice Mail Server (vmserver). Other applications are not included in the current release (1.5) and are: 1. SIP to MGCP translator 2. Policy server 3. Inet/Conference proxy server. 4. SNMP/NetMgmt 5. SIP to H323

Translator: It has IPv6 Support and a lot of subsidiary application (from the site <http://www.vovida.org>). Unfortunately, provisioning currently requires valid IPv4 addresses. For an updated list of know limitation please refer to: <http://www.vovida.org/downloads/vocal/1.5.0/doc/LIMITATIONS.txt>.

Overall Evaluation: It is a al-in-one solution for small-medium size business unit. It needs complementary solutions to be deployed in tandem in order to become a distributed system (the management is still centralised with no possibility of interfacing to other domains).

> Product Name: OpenMCU (H.323 Conferencing Server)

Product URL: <http://www.openh323.org/code.html>

Vendor: Open Source

Supported Protocols: H.323

Platform: Any platform where you can compile the OpenH323 Library (Linux, Windows, FreeBSD, Solaris, etc.)

Description: Operational Experience: It can accept multiple simultaneous connections. From the first four clients that get connected, it accepts audio and video capabilities, from the following ones only audio. It determines which conference is required via the 'rooms' feature and adds the call to that conference. New rooms are created automatically and there is a default room for people who do not specify a room or cannot specify a room (e.g. NetMeeting). It initiates calls from the MCU to remote endpoints and supports audio loop-back mode in a specific room (ideal for setup of audio hardware and testing network performance). No dynamic configuration is possible; once the program is started the client is configured using the command line options (only statistics on call in progress and initiating new calls is possible).

Overall Evaluation: MCU software requiring a performing hardware infrastructure in terms of shared memory it is using. High customisation in terms of parameters (video compression and quality) makes it scalable in terms of computational requirements. It is really useful in case of multipoint conferencing. A drawbacks is the one of multipoint conferencing, i.e., it does not use multicast but multiple unicast connections.

> Product Name: SIP Express Router

Product URL: <http://iptel.org/ser>

Vendor: iptel.org

Supported Protocols: SIP

Platform: POSIX-like systems

Description: SER or SIP Express Router is a very fast and flexible SIP (RFC3621) proxy server. Written entirely in C, ser can handle thousands calls per second even on low-budget hardware. A C Shell like scripting language provides full control over the server's behaviour. Its modular architecture allows only required functionality to be loaded. Currently, the following modules are available: digest authentication, CPL scripts, instant messaging, MySQL support, a presence agent, radius authentication, record routing, an SMS gateway, a Jabber gateway, a transaction module, registrar and user location.

The server has been optimised for speed and is being used on a couple of major SIP servers. One drawback might be the quite complicated configuration file. The configuration requires good knowledge of SIP.

> Product Name: AppEngine

Product URL: http://www.dynamicsoft.com/prod_sol/AppEngine/appenginev4.php

Vendor: dynamicsoft

Supported Protocols: SIP

Platform: Sun Solaris

Description: It is a SIP application server written in Java. The server implements Java SIP servlets which allow creation of even complex SIP applications called servlets. The servlets can interact with HTTP servlets as well. Good documentation is available with many examples.

> Product Name: Cisco IP/VC 3510 MCU

Product URL: http://www.cisco.com/univercd/cc/td/doc/product/ipvc/ipvc2_2/2_2mcurn.htm

Vendor: Cisco

Supported Protocols: H.323

Platform: N/A

Description: This MCU is an older product, identical to the RADVISION OnLAN MCU, but OEMed by Cisco and currently not supported any more, as it has been replaced by the 3511 MCU. The 3510 is capable of connecting twelve participants at 384Kbps each, or a combination of conferences at different rates and different numbers of participants. The default hardware does not provide audio/video transcoding between participants, so conference settings must be matched by all. Continuous presence is supported, but with asymmetric video rates, i.e., each participant sends 384Kbps video but receives 4x384Kbps back from the MCU, for viewing four participating sites simultaneously.

Check the IP/VC Products page at

http://www.cisco.com/univercd/cc/td/doc/product/ipvc/ipvc2_2/2_2mcurn.htm

> Product Name: Cisco MCM Gatekeeper

Product URL:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvfax_c/vvf323gk.htm

Vendor: Cisco

Supported Protocols: H.323

Platform: Cisco router - IOS based

Description: The Cisco Multimedia Conference Manager (MCM) is the name of the gatekeeper product bundled in special IOS feature sets (H.323 feature set) and can be installed on most Cisco routers that may be performing as gateways at the same time. The MCM is mostly geared towards VoIP and in supporting basic H.323 interoperability, but it has a number of extra Cisco-proprietary features as well. It can also act as an H.323 proxy for serving H.323 clients behind firewalls. The API that has been developed by Cisco allows extensive control of gatekeeper events by an external application, but it requires significant development effort to bear fruit.

Check the Cisco Gatekeeper External Interface Reference page at

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/rel_docs/gktmp4_1/index.htm

B. 4 Gateways

> Product Name: OpenISDN (H.323 Call Generator)

Product URL: http://www.gae.ucm.es/~openisdngw/home_en.php

Vendor: Open Source

Supported Protocols: H.323

Platform: Any platform where you can compile the OpenH323 Library (Linux, Windows, FreeBSD, Solaris, etc.)

Description: Operational Experience: It requires ISDN cards to be properly installed and configured on the local machine in order to make connections with the ISDN. It works only with ISDN lines (no PSTN support) managing n calls simultaneously, as many as the ISDN channels available. The gatekeeper can be in a well-known IP address or it could be discovered in the network with broadcast RAS. It gives information about the call progress state to the user of the Switched Circuit Network that calls to the Gateway. This information is made with tones similar to those sent by the telephone offices. Support and development has now stopped and it requires a special old version of OpenH323 library to compile. No dynamic configuration is possible; once the program is started the client is configured using the command line options.

Overall Evaluation: It is a simple H.323/ISDN Gateway. It needs to be better investigated for complete H.320 compatibility for ISDN conferencing. Right now it seems to be only an audio gateway.

> **Product Name: Asterisk Open Source PBX**

Product URL: <http://www.asterisk.org/>

Vendor: Digium

Supported Protocols: SIP, H.323

Platform: N/A

Description: Asterisk is an Open Source, full featured hybrid TDM and VoIP PBX and IVR platform. It allows you to seamlessly integrate TDM (T1, PRI, FXS, FXO) and VoIP (IAX, SIP, H.323) technologies in a single PBX while providing full IVR functionality through any scripting language available on Linux.

> **Product Name: Cisco IP/VC 3525 PRI Gateway**

Product URL: http://www.cisco.com/univercd/cc/td/doc/product/pvc/ipvc2_2/2_2prirn.htm

Vendor: Cisco

Supported Protocols: H.323

Platform: N/A

Description: This H.320 to H.323 Gateway is an older product, identical to the RADVISION OnLAN Gateway, but OEMed by Cisco and currently not supported any more, as it has been replaced by the 3526 Gateway. The 3525 is capable gatewaying 16 voice channels, or 8 128Kbps participants with H.261 video, or a combination of other rates for multiple BRI bonding. The default hardware does not provide audio/video transcoding, but there existed a hardware add-on for audio transcoding. Check the IP/VC Products page at http://www.cisco.com/univercd/cc/td/doc/product/ipvc/ipvc2_2/2_2mcurn.htm

B. 5 Testing

> **Product Name: CallGen323 (H.323 Call Generator)**

Product URL: <http://www.openh323.org/code.html>

Vendor: Open Source

Supported Protocols: H.323

Platform: Any platform where you can compile the OpenH323 Library (Linux, Windows, FreeBSD, Solaris, etc.)

Description: Operational Experience: It can make and receive an exact number of calls, adjust the delay between each batch of calls and set the number of batches to repeat. It only produces signalling traffic (no audio data traffic). Support and development has now stopped and it requires special old version of a OpenH323 library to compile. No dynamic configuration is possible; once the program is started the client is configured using the command line options.

Overall Evaluation: It is a simple H.323 Call Generator. It is very customisable using a number of parameters. It is really useful in testing environments where servers need to be tested under stress. Drawbacks are static configuration, no dynamic management and limited support.

> **Product Name: sipsak**

Product URL: <http://sipsak.berlios.de/>

Vendor: iptel.org

Supported Protocols: SIP

Platform: N/A

Description: Free Diagnostic and Stress Utility. sipsak is a simple utility that can be used to test various functions of a SIP server. It includes proxy, registrar and digest authentication tests. It can also generate a load of SIP messages to stress a server.

> **Product Name: SIPStone**

Product URL: <http://www.sipstone.org>

Vendor: Columbia University and Ubiquity

Supported Protocols: SIP

Platform: N/A

Description: Currently, this is a draft about measuring SIP performance <http://www.sipstone.com/>. A measurement tool is available from Columbia University - see <http://www.cs.columbia.edu/IRT/cinema/sipstone/>
See SIPstone mailing list for a discussion.

B. 6. Miscellaneous

> **Product Name: OpenAM (H.323 Answering Machine)**

Product URL: <http://www.openh323.org/code.html>

Vendor: Open Source

Supported Protocols: H.323

Platform: Any platform where you can compile the OpenH323 Library (Linux, Windows, FreeBSD, Solaris, etc.)

Description: Operational Experience: It can accept multiple connections simultaneously and runs

a user-defined program after each call, which can be used to automatically send the recorded message as a MIME-encoded e-mail attachment to a known e-mail address. If the recorded message is encoded using G.723.1 codec, it requires equipping with the PC with additional cards (Quicknet). No dynamic configuration is possible; once the program is started, the client is configured using the command line options.

Overall Evaluation: It is a simple answering machine using the H.323 protocol. It is really useful in unified messaging scenarios. It needs to operate in an environment where supplementary services are implemented. Drawbacks are static configuration and no dynamic management.

> Product Name: Yxa

Product URL: <http://www.stacken.kth.se/projekt/yxa/>

Vendor: Open Source

Supported Protocols: SIP

Platform: Any platform where you can use Erlang programming language (Linux, Windows, FreeBSD, Solaris, etc.)

Description: Operational Experience: Yxa is a bunch of library-like functions for receiving, processing and sending SIP messages, and a couple of small programs that can do various things. The operational experience is poor right now because Yxa is not widely known (even if it is gaining popularity). It is a SIP server open source software written in Erlang (a programming language from Ericsson with open source releases). Basically, it has built-in some software for performing a number of functionalities. One of the main applications is the incoming proxy; it can handle REGISTER requests and authorise different users to make calls to different classes of PSTN numbers. It can proxy requests from UACs to other parts of the Yxa system, relay requests to remote servers/domains Routing features, do ENUM lookups of things that looks like E.164 phone numbers and do lookup addresses in LDAP.

Overall Evaluation: The project just produced some nice results. No release has been made yet and downloading is done only through CVS. A mailing list is available for questions and inquiries.