

# APPLIED MULTIPROTOCOL ROUTING IN IP TELEPHONY

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**Abstract.** The aim of the project was to create a multi-protocol system using SIP, H.323 and MGCP standards, which would ensure routing to various types of VoIP networks. The priority was to provide multi-protocol support to SIP and H.323 signalling and the support of the routing using the ENUM standard which has recently passed from the trial phase into full operation in the Czech Republic. The document describes the system's architecture and the components used. It also briefly describes ENUM.

## Keywords

SIP, MGCP, H.323, ENUM, VoIP, routing.

## 1. Introduction

The multi-protocol routing system was designed and implemented in CESNET and consists of five key components and enables connecting voice gateways, end users and peering with other IP telephony providers. The whole system was incorporated into the current IP telephony system and its functionality was tested in trial operation. Although it is not publicly accessible, it demonstrates the variety of connection by means of different equipment by various producers using various protocols; this is project's key contribution.

## 2. Architecture of the Open Multi-Protocol Routing System

The key component of the system is hipath 8000 ("h8k") - a SIP soft switch developed and produced by Siemens. In the course of the trial operation in the Czech Republic, CESNET managed to obtain h8k software, its installation and workshop held in Vienna free of charge. All we had to buy were the servers. We would like to thank the management of the Siemens Division COM for their support and the engineers who participated in installation and debugging of the configuration. The h8k used by CESNET is built on Suse Linux and IBM x346 HW

platform. Its core consists of two IBM x346 servers operating in a fully redundant mode. H8k is complemented by one server for management and by one media server.

The system's primary protocol is SIP, see [1], [2], [5], and the communication with H.323 is ensured by means of a protocol gateway. The system is managed through a web interface. The implemented h8k supports the Privacy Mechanism for the Session Initiation Protocol and IPsec security, which is a means of communication security which we have not yet tested since it was not available under the open-source solutions used.

Figure below shows interconnection of the components and routing prefixes.

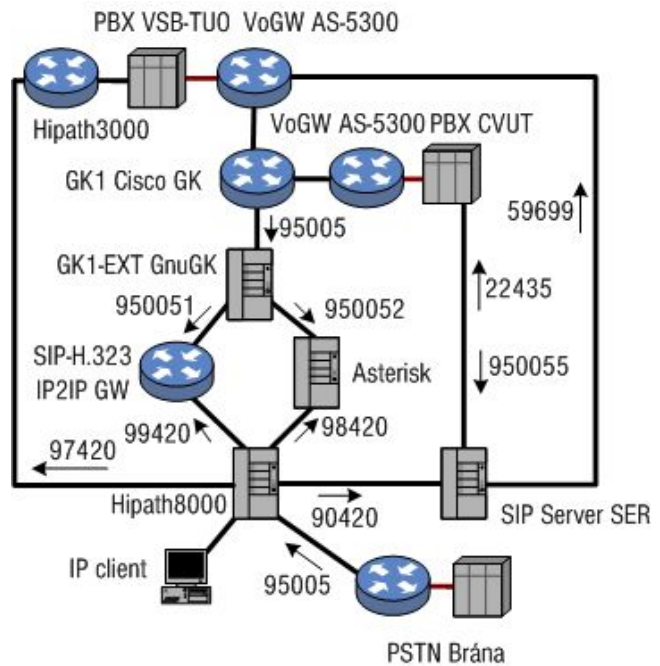


Fig. 1: Scheme of the network for operation testing.

The system consists of following components:

- soft switch hipath8000 (Siemens),
- translation gateway H.323-SIP (IP2IP GW Cisco),
- SIP Express Router (open-source for Linux),

- Asterisk used as the translation gateway H.323-SIP-MGCP (open-source for Linux),
- GNU Gatekeeper (open-source for Linux).

The translation between SIP and H.323 is currently carried out at the Cisco IP2IP gateway but we are also looking for different solutions based on open-source and we are testing Asterisk as the translation gateway. We are going to run the IP2IP gateway in parallel with Asterisk until we have verified the full functionality of the translation on Asterisk. The above mentioned components are part of the architecture which provides services to equipment (gateways, IP phones) connected to it. In order to verify whether these are compatible, we drafted a list of devices based on which the soft switch acceptance protocol was designed. The devices were chosen from the following four categories:

- translation protocol gateway,
- SIP Proxy,
- voice gateway,
- IP telephones.

The test routing prefixes are mapped statically in order to simplify the verification of component accessibility and to make it more transparent. The critical spot is the SIP-H.323 translation gateway. The prefix enables routing the call to a particular translation gateway in both directions. Later on, we focused on ENUM, i.e. mapping of E.164 number to URI via DNS. At present, we have a fully functional and verified ENUM on the SIP Express Router. Besides the e164.arpa public tree we also use the access into nrenum.net and the cesnet.cz private tree.

We ensured the delegation of most prefixes of the members of the CESNET association in the ENUM system. Our experience and achieved results enabled us to participate in developing the new communication architecture of the European infrastructure of national research networks under TF-EFC (Task Force Enhanced Communication Services) of TERENA association.

In our system, SER, h8k, ASTERISK and GnuGK support ENUM. We set up ENUM on SER, GnuGK and Asterisk. The SER and GnuGK servers are also used as border components in the CESNET VoIP network for communication with other VoIP networks and therefore ENUM NAPTR records in DNS are directed to them. Unfortunately, Enum did not operate properly in this version of h8k. The following chapters describe individual system components.

### 3. Hipath 8000

Hipath8000 is a soft switch based on SIP scalable to serve up to 100 000 users. Its redundancy is ensured by interconnection of two identical servers. These servers are connected through multiple redundant 1Gbps interfaces. Figure 2 shows the interconnection of servers, we can see the physical interconnection of interfaces for individual networks (Signalling, Billing, Cluster Interconnect and Administration). The redundancy ensures an uninterrupted operation in case one of the servers breaks down and in theory also a smooth upgrade without outage.

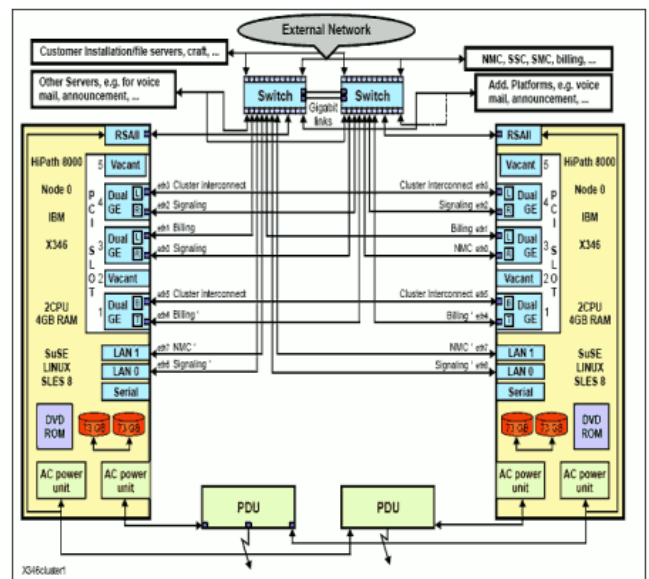


Fig. 2: Interconnection of IBM x346 servers.

H8k was installed on two IBM servers x346. Each of the servers has seven Gigabit ethernet interfaces and together they form a cluster which is subdivided into four VLANs:

- signalling,
- administration NMC/SMC,
- billing,
- interconnect.

Beside these two IBM x346 servers, another server - media server Convedia CMS-1000 is connected to the h8k system. CMS communicates with h8k using MGCP and SIP signalling and enables creating conferences and attaching audio announcements to calls. This Media server is connected to switch by means of two ports designed for:

- management,
- control and media.

The h8k system is managed by two applications running on one Primergy RX100 S3 server. These are iNMC (Network Management Control) and iSMC

(Subscriber Management Control) applications. The latter enables managing through the web. All configuration tasks can also be carried out via CLI (Command Line Interface) which runs directly on the h8k console on Suse Linux or via a secure shell. The currently installed version of h8k is 2.1. At the Vienna workshop organized for us by Siemens, we had opportunity to test the higher version which works with the Assistant. The Assistant replaces iNMC and iSMC and unifies the access to the system. The management is easier in it. However, this version was not released at the time of the installation. In the Siemens laboratories, the future development of h8k was described to us. The SS7 support is a very interesting extension of the h8k operator version. As we have been using the SS7 open-source solution on Asterisk [7], it might be possible to connect these two systems through SS7. The SS7 signalling system is used in IN (Intelligent Networks) of the telecommunications operators of both fixed and mobile networks. These operators make peering solely on SS7.

We chose Convedia CMS-1000 as the Media server for our solution. The Media server is the terminal mixing point of RTP streams. It contains the codecs that perform mixing, audio recording and playback. CMS is a stateless device. It allocates media channels when directed to it by the soft switch or by the application server but does not attempt to maintain call state. The Media Server has the following features:

- tones,
- announcements,
- DTMF collection, generation, relay,
- fax support (detect T.30 / T.38),
- conferencing (n-way, meet-me),
- lawful Intercept (CALEA) support,
- multi-lingual options,
- integration with ASR and TTS (via MRCP),
- support SIP and MGCP control,
- CSTA and VXML applications interfaces,
- record and playback options,
- SNMP Management.

H8k enables connecting SIP phones and gateways and their interconnection with SIP Proxy. H8k supports H323 but this solution is not fully compatible with H.323 standards used in CESNET2 network and will be completely removed from the later version. This is why we use translation gateways.

## 4. SIP Express Router

Another component used is the SIP Express Router (SER). SER is an open source project written in C language. It covers the function of the registrar, redirect and proxy server for communication by means of the SIP protocol. It was originally developed as a backbone SIP router with strong focus on its performance. It can serve thousands of calls per second on the two-processor PC and hundreds of calls on IPAQ. Server supports stateless and transaction statefull processing of SIP communication. This processing is controlled by a powerful script language. But the powerful language means that its configuration is rather complex. The server is modular and as such is very well extendable. As the number of users keeps growing, new modules are created. SER can be controlled remotely via FIFO or UNIX socket. SIP stack supports communication over the IP protocol, versions 4 and 6, transport protocols UDP and TCP and can also support TLS with the appropriate add-on module. The server also supports multiple user domains, aliases and ENUM. The authentication and authorization can be done via Radius or modules using memory storage or database (Mysql, Postgress). The registration information can also be stored in a SQL database. It is not necessary to use the database to run SER. The registration information are thus stored only in the memory which is fast but these records are lost upon a restart. The SER is able to help with NAT traversal (nathelper, rtpproxy, mediaproxy) and its potential could be extended by external scripts written for example in Perl. The SER is designed for Linux, BSD (NetBSD, FreeBSD) and Solaris.

The SER is used in CESNET as the access router and as a server for users. You can download the current version of the SER is available either as a package for Debian or as a zipped tar ball of source files. Afterwards, you could modify the Makefile to set up which modules will be compiled. For instance Mysql module is not compiled by default. The ser.cfg configuration file is by default located in /usr/local/etc/ser.

## 5. GnuGK

GnuGK runs in CESNET2 network on server gk1ext.cesnet.cz and is, together with the SER, one of the key elements of the system. Its configuration and functions were described in detail in the technical report [3]. Under the project, the support for ENUM has been set up. The configuration is saved in the file /etc/gatekeeper.ini, but before the GK is launched, the PWLIB\_ENUM\_PATH variable has to be set up. Once it has been tested, it can be set up in /etc/profile. We set up the ENUM queries into two trees - e164.arpa and nrenum.net. This is done by means of the following command:

```
#export
PWLIB_ENUM_PATH=e164.arpa:nrenum.net
```

The GNU Gatekeeper depends on OpenH323 library which is dependent on PWLib. ENUM requires at least PWLIB V1.8.0. The GNU GK documentation at GNU Gatekeeper site contains a mistake in the ENUM description. A quote from the gnugk documentation: "To specify your own server you have to specify an environmental variable PWLIB\_ENUM\_PATH with the address of your preferred enum servers separated by a semicolon (;)." But semicolon (;) is wrong (for Linux), the right syntax is colon (:).

Setting up the /etc/gatkeeper.ini file is quite easy. All you have to set up in the [RoutingPolicy] section is the directions which should be available via ENUM, e.g.:

```
[RoutingPolicy]
123456=enum,dns
789847=enum,dns
```

```
or default rule
[RoutingPolicy]
```

```
default=explicit,internal,neighbor,parent,enum,dns
```

The next configuration snippet shows routing of calls to h8k either via the IP2IP or Asterisk translation gateway. We describe only the part of the configuration relating to routing.

```
[RasSrv::GWPrefixes]
IP2IPGW=950051
GW-CESNET-AST=950052
[RasSrv::GWRewriteE164]
IP2IPGW=out=950051=420950051
GW-CESNET-
AST=out=950052=420950052
[RasSrv::PermanentEndpoints]
IppaddressGWAsterisk:1720=GW-
CESNET-AST;420950052
IppaddressGWIP2IP:1720=IP2IPGW;42
0950051
```

## 6. Asterisk as SIP-to-H.323 Gateway

In our project, we use Asterisk as the translation gateway between SIP and H.323 in the same way as Cisco IP2IP Gateway. Asterisk is designed as a multi-protocol PBX. SIP and H.323 are only two of the various protocols which Asterisk is able to use and the translation between these two protocols is just one of possible ways to use Asterisk. As Asterisk is not designed directly for such a translation, the configuration is broken down into several files [4]. We will therefore move directly to the configuration of the significant parts used in the project.

The configuration of the SIP channel is stored in the sip.conf file. All SIP clients or superior servers are set up

there. For our purposes, we configured only few clients (IPphones). The configuration is divided into several sections: the general section which contains the initial settings for all / non authenticated clients and individual sections which describe peers.

```
[general]
context=guest ; Default context for
incoming calls
port=5060 ; UDP Port to bind to (SIP
standard port is 5060)
bindaddr=0.0.0.0 ; IP address to bind
to (0.0.0.0 binds to all)
srvlookup=yes ; Enable DNS SRV
lookups on outbound calls
tos=lowdelay
owdelay,throughput,reliability,mincos
t,none
defaultexpiry=400 ; Default length
of incoming/outoing registration
videosupport=yes ; Turn on support
for SIP video
disallow=all ; First disallow all
codecs
```

```
[general]
allow=h263
allow=h263p
rtptimeout=60 ; Terminate call if 60
seconds of no RTP activity
rtpholdtimeout=300 ; Terminate call
if 300 seconds of no RTP activity
```

```
;Number 950052 11
; all calls from this number will nbe
routed to context sip.
[950052 11]
username=950052 11
secret=nejakeheslo
callerid=Pepa Nos
type=friend
context=sip
host=dynamic
canreinvite=no
qualify=6000
nat=yes
```

The ooh323.conf file describes the setting of the ooh323 channel. Asterisk supports more H.323 channels: the original asterisk's h323, oh323 built on the openh323 stack and Objective Open H.323 (ooh323c). For our purpose, we used ooh323c because it is written in purely in C and does not, like the oh323, depend on other external libraries. Thanks to these qualities it is more convenient for the possible embedding. The next configuration file is extension.ael. The new syntax of dial plan in Asterisk (Asterisk Extensions Language) is very synoptic. Unlike the extensions.conf, AEL has a higher-level syntax. We used AEL to define the enum macro according to a contribution found at the voip-info.org Wiki. Asterisk supports ENUM only at a low level (enumlookup

function). Anything more complicated depends on user functions, but it gives the ENUM lookup a substantial variety. The initial configuration of ENUM in `enum.conf` is not used in this case and zones are defined directly within the macro. It is necessary to interconnect these channels so that the call from the first protocol is routed to the second as well as in the opposite direction which is done in `extensions.conf`. We selected ENUM because it is much easier, more legible and clearer than Asterisk itself.

Now we need to interconnect everything, the authenticated clients from SIP will call via context `sip` and non -authenticated via `guest` (see `sip.conf`). The process for H.323 is rather more complicated because we only support anonymous users that can be routed based on their IP address only. We did not use this option and we route everything from H.323 to the `h323` context. We should stress that one should not rely on the order of lines in individual contexts. The sole truly reliable element is the order of include directives. Further we note that the lines written directly in the contexts are assessed first and then the includes.

## 7. IP2IP Gateway

We have used Cisco 2651XM as the translation gateway with the `c2600-adventerprisek9_ivs-mz.124-3b` image. The configuration was done according to the documentation. The following syntax in configuration allows the translation:

```
voice service voip
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
```

In the SIP direction, the SIP server is set as the `session-target`. The SIP server is set up in `sip-ua` section and resolved via DNS:

```
dial-peer voice 211 voip
destination-pattern 950051...
session target sip-server
```

In the H.323 direction, RAS is set as the `session-target`. The setting for `ras` can be found at the Fast Ethernet interface in which we configured the registration of gateway at `gk1ext.cesnet.cz`.

```
dial-peer voice 1 voip
destination-pattern 420.....
session target ras
```

At IP2IP GW we also use translation of numbers and codec preferences which is set up in the Voice class codec section and Translation rules.

## 8. ENUM

ENUM plays an important role in the interconnection of VoIP islands. Its base function could be described as the mapping of the space of E.164 telephone numbers into the space of URIs (uniform resource identifications). The most often IP telephony URIs used begin with `sip:`, `h323:` or eventually `tel:`. The core ENUM functionality has been described quite frequently and therefore we will not deal with it further. Let us only remind that the DNS system is used to distribute information (translation rules), in particular NAPTR of records.

An example NAPTR of the record for `9.8.2.5.5.3.4.2.2.0.2.4.e164.arpa:`

```
IN NAPTR 100 50 u "E2U+sip"
"!^\\+(.*)$!sip:\\1@cesnet.cz!" .
IN NAPTR 200 50 u "E2U+h323"
"!^\\+(.*)$!h323:\\1@gk1ext.cesnet.cz!" .
```

These rules transform E.164 number `+420224355289` into URIs `sip:420224355289@cesnet.cz` and `sip:420224355289@gk1ext.cesnet.cz`

The DNS system is quite suitable for this purpose thanks to its stability and the distributed approach. In addition, the mechanisms resolving the address for the target element also use DNS. Besides the common translation of names to IP addresses the SRV records can also be used. SRV records publish the places of the operator of individual services. The following SRV records show service points for the SIP protocol using transport protocols TCP and UDP for the `cesnet.cz` domain:

```
_sip._udp.cesnet.cz SRV 100 10 5060
cyrus.cesnet.cz.
_sip._tcp.cesnet.cz SRV 100 10 5060
cyrus.cesnet.cz.
```

The disadvantage of using DNS is a rather low security level because DNSSEC is not that widely used and the transmitted information cannot be protected against forgery. The system is designed so that anyone can ask for information. The way in which various answers are created based on the address of the participant who poses the question is rather complicated, in particular where recursive queries are used.

The benefit of the system is that the owner of the telephone number controls information that is made public about his number. The most frequent has so far been the exchange of routing information (prefixes and addresses of border elements) while interconnecting various islands. When the information has changed, it had to be distributed to all participants and their systems adjusted accordingly.

The probability that a mistake will occur increases with a larger number of participants. When every participant only publishes own information, the system is much flexible.

We stored prefixes of operators which we are connected to us in a private tree. All NAPTR records about them are stored directly in a single zone, i.e. not delegated. This not only enables every participating operator to transfer the whole zone to himself and thus obtain faster responses, reduces the likelihood that the records will change during the transport but it reduces the possibly for every operator to manage his own records in his name server. We try to remove this disadvantage by designing such a zone management system in which the participants could also publish contact persons' information and testing numbers which facilitates solving possible issues. The system consists of a web interface built upon MySQL database, a functionality which ensures that the zone is generated and of DNS server BIND.

## 9. Conclusion

The output of our work is the implementation of a multi-protocol system working with standards SIP, H.323 and MGCP which uses both the static routing and ENUM. We tested the functionality of the two-way translation between H.323 and SIP by means of a load test using the IP2IP gateway. Moreover, we added a new feature into the system based on Asterisk. We explored open-source solutions and used the GNU GK, SIP Express Router and Asterisk open projects. The core of the tested solution consists of the Hipath8000 soft switch produced by Siemens which supports several advanced functions and enables interconnecting advanced enterprise or operator oriented system with our largely open source systems and provides the background for such cooperation. ENUM ensures the openness and availability of the connection. The next phase is to build up a system of trust on the interconnected networks of IP telephony islands.

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