Two Standards: H.323 / SIP
Bridging both worlds

João Pereira - FCCN - Portugal
Presentation

- FCCN - Portuguese NREN (RCTS) administrator and a research unit;
- We’ve been using H.323 videoconferencing since 2000;
- GDS National Gatekeeper Administrator since 2001;
- VoIP pilot at FCCN and in some institutions;
- Only standard protocols! (H.323, SIP);
- The need for bridging both worlds.
Summary

- Before we start
- Protocols
- Implementation Differences - Signaling and Streaming
- Addressing
- Tools / Applications
- System Architecture
- Bridging
Before we start

• Research about the available VoIP protocols;
• Search for the best servers to support them;
• Implement logging, accounting, authentication, authorization, ...;
• Dialing methods (GDS, ENUM, UMMAP, URI);
• Architectural definitions for RCTS institutions;
• Support for the existing and new network services
  (E-u, IPv6, PT-GK, Estudios, ...)
Protocols - H.323

The ITU Protocol

Advantages

• Widely adopted and deployed initially;
• Designed from the outset with multimedia communications over IP in mind;
• Provides a more precise and detailed specification of voice and multimedia functionality
• Perfect solution for real-time multimedia communication over packet-based networks;
• Umbrella protocol with a lot of protocols inside;
• Straightforward interworking with the PSTN;

Disadvantages

• NAT and firewall restrictions;
• Hierarchical structure based on the old PSTN world;
• ASN.1 syntax for coding H.323 messages is more complex;
• Range of applications because of its restricted scope;
Protocols - SIP

IETF protocol for VoIP and other text and multimedia sessions.

Advantages

• Widely adopted and deployed today (and growing...);

• Flexible, as we can add more features;

• Messages consist of headers and a message body;

• Session-layer transaction protocol;

• Provides advanced signaling and control functionality;

• Standardized on the transaction level and not on the supplementary service level;

Disadvantages

• Suffer from NAT and firewall restrictions
Implementation Differences

H.323
- Gatekeeper / MCU / Gateway
- "Old World" - complex, deterministic and vertical
- Designed as a control protocol suite with the focus on multimedia applications;
- Top down approach
- Describes a complete framework and detailed protocols;

SIP
- Proxy server, registrar, ConfServer, Gateway
- "New World" - a relative of Internet protocols - simple, open and horizontal
- Designed as a general transaction protocol for setup and tear down of generic sessions. Voice and multimedia are only possible example applications
- Bottom-up approach
- Formed by combination of generic modules;
Signaling

H.323
- H.245 is the Control Protocol for Multimedia Communication
- Binary signaling (ASN.1);
- Possibilities of delay;
- RAS (Registration, Admission and Status) signaling functions are required for endpoint registration, admission control, and address resolution;
- Uses UDP in port 1719 (usually);
- Basic call setup / Fast connect;

SIP
- Text signaling, much like HTTP or SMTP;
- Minimal delay - simplified signalling scheme makes it faster
- Uses TCP in port 5060 (usually);
- SIP-INVITE transaction
Streaming

H.245 Control signal

- Used for exchanging terminal capabilities and creation of media channels;
- RTSP is the IETF protocol for controlling streaming media servers;
- RTSP URIs are hierarchical and based on a file mode;
- RTSP has just one-directional media.

SDP Call Control

- SDP is a text-based media-description format that can be carried in the SIP message body;
- The URIs are email-like;
- Usually initiates bidirectional sessions;
Addressing

H.323

- Gatekeeper hierarchy (ex: GDS, E.164);
- Only in a recent Annex became possible to dial with URL;
- Addressing scheme doesn't scale well;

SIP

- Hierarchical URL style addressing scheme that scales;
- Allow the use of SRV Records in the DNS;
- URLs can be embedded in web browsers and email tools;
Other Services

- Authentication
- Authorization
- Accounting
- VoiceMail
- Gateways
- Value Added Services
- Proxy
- MCU
- GDS
Tools

SIP Express Router
- Implements SIP AAA
- Works as proxy or just router
- Supports RADIUS, PGSQL, MySQL
- SERweb for client web admin
- RTPproxy to use as proxy

Asterisk
- SIP and H.323 support
- Has also support for other protocols
- Complete PBX with value added services (VoiceMail on email, call conference, music for call on hold)
- PSTN Gateway

Open H.323 GNU Gatekeeper
- Direct Q.931 routing, Q.931 and H.245 routing.
- AAA / RADIUS and SQL support
- H.350 (LDAP)
Why not just Asterisk?

- It doesn’t have all the gatekeeper functionalities needed;
- Isn’t a fully functional SIP server and doesn’t support thousands of clients as SER;
- It’s not a SIP proxy, but a SIP registrar and location server;
- It acts as a SIP agent, placed between the calls;
- This server dependency doesn’t fit the distributed SIP architecture;
- A SIP Proxy is never the end of a call, and should never deal with the media stream as Asterisk does;
System Architecture

- **FCCN Gatekeeper**
- **Asterisk**
- **SER**
- **VoiceMail**
- **MCU**
- **confserver.fccn.pt**

- **H.323** terminals
- **SIP** terminals
- **WiFi SIP terminals**

- Gateway, value added PBX services, ...
- Extendable to other VoIP protocols
System Architecture

FCCN Gatekeeper

H.323 World

FCCN Gatekeeper

Data Base

H.323 Terminals

Accounting, Authorization, Authentication

Gateway, value added PBX services, ...

Asterisk

H.323 Terminals

Siemens VoIP Terminals

Analogic and Digital Terminals

Inbound / Outbound 4 BRI (8 channels)

PSTN PBX

SIP World

SIP Terminals

SIP Softphones

WIFI SIP Terminals

VoiceMail

MCU
dconfserver.fccn.pt

Extendable to other VoIP protocols

Extendable to other VoIP protocols

Outbound / Inbound

PSTN Outbound / Inbound

Analogic and Digital Terminals
Dial Plan

PSTN

PBX

Asterisk

GNU Gatekeeper

SER
Configuration

SIP Express Router
- Has one configuration file: ser.cfg

GNU Gatekeeper
- Static configuration in a file called gnugk.ini
- It’s a text file with blocs defined by tags

Asterisk
- The dial plan is configured in extensions.conf
- The channels are configured in specific files: (sip.conf, oh323.conf,..)
Bridging

SIP client

SIP server (host:port)
Username
Password
Domain (fccn.pt)

Registrar
DB
Dial Plan
Routing
Proxy

Dial Plan
Routing
Proxy

Registrar
DB
Routing
Proxy

Gatekeeper (host:port)
Username
E.164

H.323 client
Bridging

SIP client

SIP server (host:port)
- Username
- Password
- Domain (fccn.pt)

Registrar
- DB
- Routing
- Dial Plan
- Proxy

H.323 client

Gatekeeper (host:port)
- Username
- E.164

if (method=="REGISTER") {
  if (!is_from_local()) {
    sl_send_reply("403", "Unknown Domain");
    break;
  };
  if (!www_authorize("fccn.pt", "user")) {
    www_challenge("fccn.pt", "0");
    break;
  };
  if (!save("location")) {
    sl_reply_error();
    }; break;
};

DB

select password,domn,uuid,rpid from utilizador where username='joao.pereira'

select grp from grp where username='joao.pereira' AND grp='disabled'
Bridging

SIP client

SIP server (host:port)
Username
Password
Domain (fccn.pt)

Registrar
DB
Routing
Dial Plan
Proxy

Asterisk

Gatekeeper (host:port)
Username
E.164

H.323 client

Dial Plan

if (uri=~"^72[0-9{3}]"){
  if (isflagset(28)) {flag28 means you have external PSTN permissions
    rewritehost("asterisk.fccn.pt:1720");
  } else {
    sl_send_reply("503", "Call not allowed.");
  }
} break;


Bridging

SIP client

SIP server (host:port)
Username
Password
Domain (fccn.pt)

Registrar
DB
Routing
Dial Plan
Proxy

Gatekeeper (host:port)
Username
E.164

H.323 client

Proxy
if (method=="INVITE" || method=="ACK") {
  force_rtp_proxy();
}

if (method=="INVITE" || method=="ACK") {
  unforce_rtp_proxy();
}
Bridging

SIP client

SIP server (host:port)
  Username
  Password
  Domain (fccn.pt)

Gatekeeper (host:port)
  Username

Registrar

Proxy

DB

Routing

Registrar

[SQLPasswordAuth]
Driver=PostgreSQL
Host=193.136.252.51:5432
Database=gnugk_database
Username=db_username
Password=db_password
CacheTimeout=300
Query = SELECT password FROM user WHERE username = '%1' AND active = true;

Proxy

[Proxy]
Enable=1
T120PortRange=40000-40999
RTPPortRange=50000-59999
Bridging

SIP client → SIP server (host:port) → H.323 client

- Username
- Password
- Domain (fccn.pt)

Routing

[RoutingPolicy]
- h323_ID=dns,internal
- 72=internal
- 74=neighbor
- Default=internal,neighbor,parent

Routing

[Neighbor::GWasterisk]
- GatekeeperIdentifier=GWasterisk
- Host=193.136.252.51:1720
- SendPrefixes=0,74
- AcceptPrefixes=0,72
- ForwardLRQ=never

DB

SELECT rules.rule FROM rules, group, user, group_rules_param WHERE (rules.id_rule = group_rules_param.rule AND group.id_group = group_rules_param.group AND user.group = group.id_group AND user.username = 'joao.pereira')
**Bridging**

SIP client

- SIP server (host:port)
- Username
- Password
- Domain (fccn.pt)

**Dial Plan - extensions.conf**

**[default]**

- `dial ${EXTEN}@193.136.252.51,30,r` if the number dialled is 7, send it to the default context

- `exten => _74XXX,1,Dial(SIP/${EXTEN}@193.136.252.51,30,r)
  exten => _72XXX,1,Dial(H323/${EXTEN}@193.136.252.51,30,r)`

- if the number dialled has five digits and starts with 74 (or 72), send to the SIP channel

**[from-sip]**

- `dial CAPI/12345678:b${EXTEN}|90` if the number dialled comes from SIP and has three or more digits, send it to the ISDN line

- `exten => _XXX,1,Dial,CAPI/12345678:b${EXTEN}|90`

- `exten => _0XXXXXXXXX,1,Dial,CAPI/12345678:b${EXTEN}|90` if the number dialled has 0 plus nine digits, send it to the ISDN line

**[from-h.323]**

- `exten => _XXX,1,Dial,CAPI/12345678:b${EXTEN}|90`

- `exten => _0XXXXXXXXX,1,Dial,CAPI/12345678:b${EXTEN}|90` if the number dialled comes from H.323 and has three or more digits, send it to the ISDN line

**[from-pstn-pbx]**

- `exten => s,1,Answer` if the number dialled is 7, send it to the default context

- `exten => 7,1,DISA(no-password|default)` answers the call, if the number dialled is 7, send it to the default context

H.323 client

- Gatekeeper (host:port)
- Username
- E.164
Bridging

SIP client

SIP server (host:port)
Username
Password
Domain (fccn.pt)

H.323 client

Gatekeeper (host:port)
Username
E.164

Dial Plan
Routing

in sip.conf
[193.136.252.51]
type=peer
username=193.136.252.51:5060
host=193.136.252.51
context=from-sip
canreinvite=no
insecure=very

in oh323.conf
[general]
gatekeeper=193.136.252.51
gatekeeperPassword=fccnasterisk
context=from-h323
Conclusion

The system is designed for H.323 and SIP, but is upgradable to other protocols;

This system is already in production in FCCN;

The tests with Universities have already started and will be concluded in November;

A lot of documentation was produced, and much more will be made until the end of the tests;

Our goal is to allow our Universities to implement our dual system and join us in a big VoIP network.
Links

www.iptel.org

www.gnugk.org

www.asterisk.org

www.voip-info.org

www.terena.nl/library/IPTELEPHONYCOOKBOOK/

Any questions?

João Pereira

sip:joao.pereira@fccn.pt