Development of SIP-H.323 Gateway Project

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v2

SIP-H.323 Gateway project

- Motivation
  - Large deployment base of H.323 terminals (over 2.9 million calls placed over the AARNet H.323 VoIP network in 2004)
  - Will SIP replace H.323? Will H.323 die instantly? Will these protocols run in parallel?
- As a pre-requisite to using SIP, we need either an endpoint that has a SIP protocol stack, or
- a means to translate the SIP messages to the protocol that is natively supported

Ways to achieve protocol conversion

We have used three products to convert SIP to H.323 and vice-versa

Asterisk – Open source PBX (http://www.asterisk.org/)

Cisco Multiservice IP-to-IP Gateway (http://www.cisco.com)

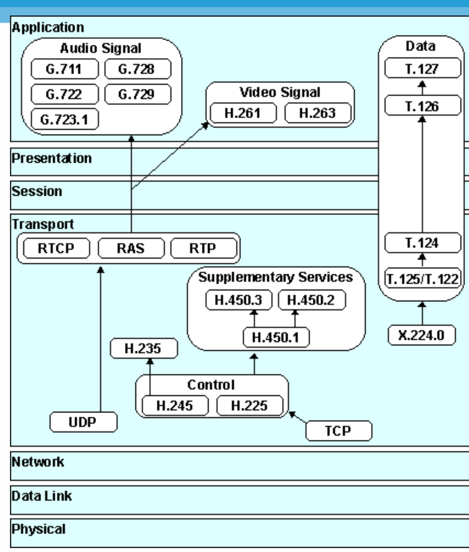
“Back to Back” gateway (AS5300 with 2xE1)

REMEMBER: We’re only converting the call signalling, RTP is the same. (media transcoding is not required)

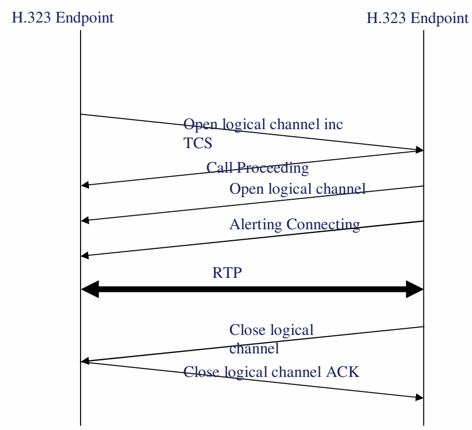
*Other products becoming available eg: Yate

http://yate.null.ro/pmwiki/ - Yet another Telephony engine

H.323 protocol stack – see, it’s confusing



H.323 protocol flow



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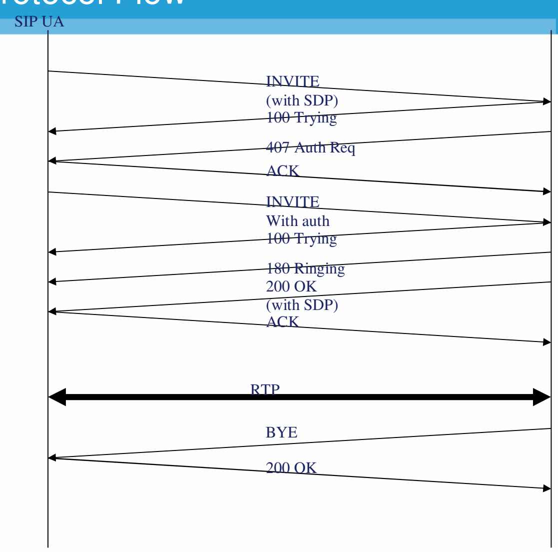
SIP Protocol Flow



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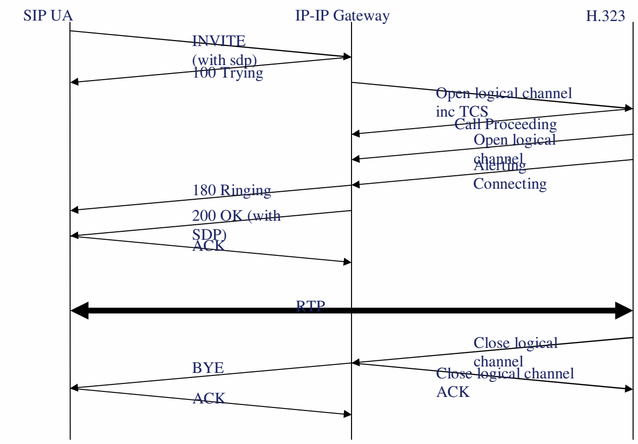
SIP to H.323 Protocol Flow



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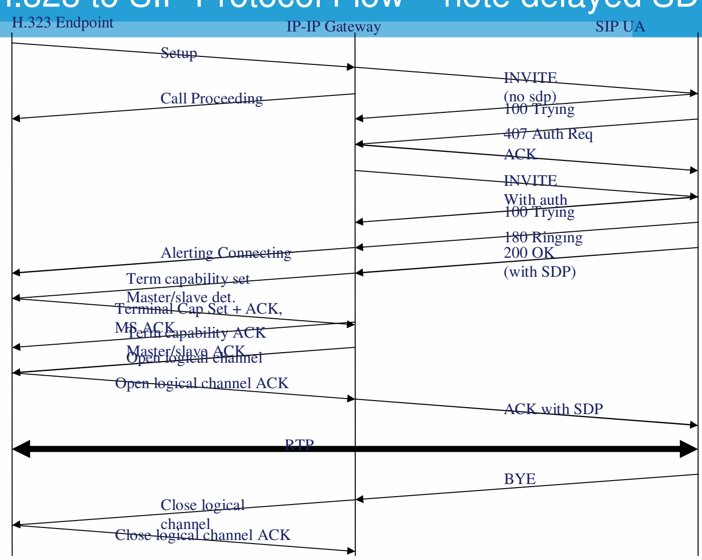
H.323 to SIP Protocol Flow – note delayed SDP



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Cisco

- SIP-H.323 and H.323-SIP support was introduced in IOS release 12.3(11)T
- Supports Voice and Video
- Not a big performance impact, as no media transcoding is involved. At worst case, just copy the media stream from input buffer to output buffer
- Costs money (2 cisco routers, 1or 2 ‘expensive’ software license)

Asterisk

- Asterisk “likes” media to flow-through, rather than flow-around.
- Supports Voice and Video
- Asterisk **must** know about the RTP protocol before it will allow calls to be setup
- H.323 is not a supported channel type “out of the box”. You need to compile a separate channel driver – version-ing IS a problem
- Free

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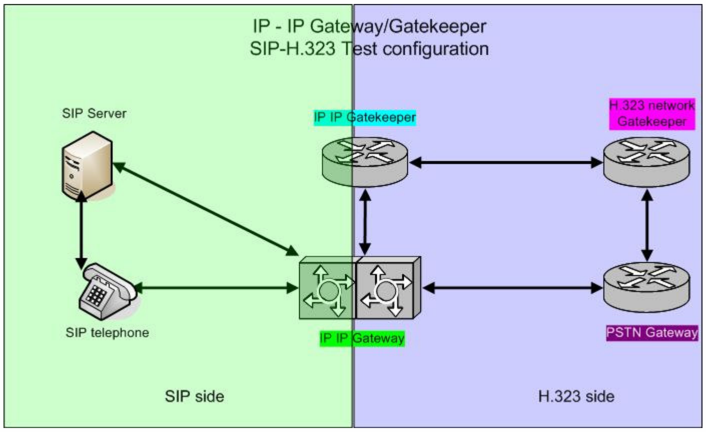
SIP-H.323 Gateway Network Architecture (Cisco IP-IP Gateway)



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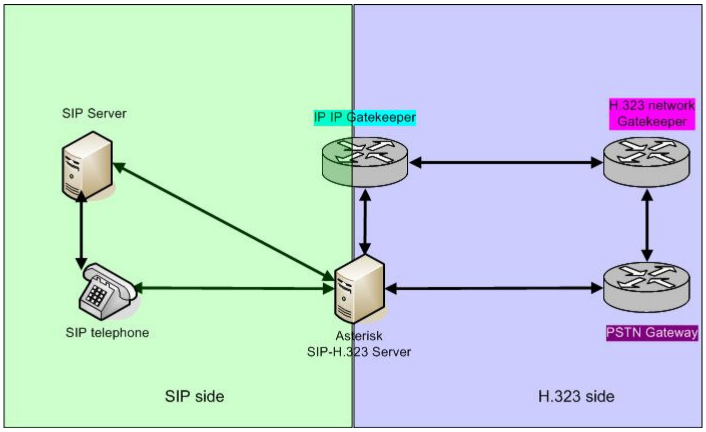
SIP-H.323 Gateway Network Architecture (Asterisk Server)



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Codec Negotiation

- Currently both IP-IP gateway and Asterisk products have issues selecting the correct codec
- Back to back gateway does not have this issue, as it converts to ISDN first

Dial plan issues

- Desire for “A” party not required to be aware of “B” party’s technology
- Other option to have an “access code” – this requires either the user to know the B party’s technology type, or the SIP proxy or gatekeeper to insert a routing prefix

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Product comparisons

- Asterisk: Was never designed to do what we are asking of it. It was designed as a soft-pabx, with small-medium number of telephones attached, and small numbers of trunk connections

- H.323 support is an “add on”, and is not supported all that well.
- Back-to-back is not elegant, with some latency introduced, and possible voice quality degradation

Product Comparisons

- Cisco IP-IP Gateway/Gatekeeper
- Was originally designed as a H.323 – H.323 product.
- Has very good H.323 support
- Was not designed as a softswitch, so does not support large “routing tables”. Routing is achieved by adding dial-peers. Each dial peer requires approximately 6kb of memory.
- While this allows 5000 theoretical dial peers on an “average” router, this becomes unmanageable

SIP and H.323 Feature compatibility

- Obviously, some features exist only in one protocol, but not in the other. Regardless of how good the protocol converter is, if features don't exist in the protocol standard, they can't be supported)
- Examples include:
  - Security
  - Presence
  - Support for different types of sessions (IM)

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SIP Competition

AARNet crystal ball competition:

When do you believe SIP will replace H.323 in your institution?

Write your answer and submit to the committee.

Results published at QUESTnet 2005.

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Acknowledgements

- We wish to acknowledge the assistance of Cisco Systems with assistance configuring and fault-finding the IP-IP gateway/gatekeeper
- We wish to acknowledge the assistance of Brendan Beverage, from Broadreach Services (02) 8270 1000 for assistance configuring the Asterisk product

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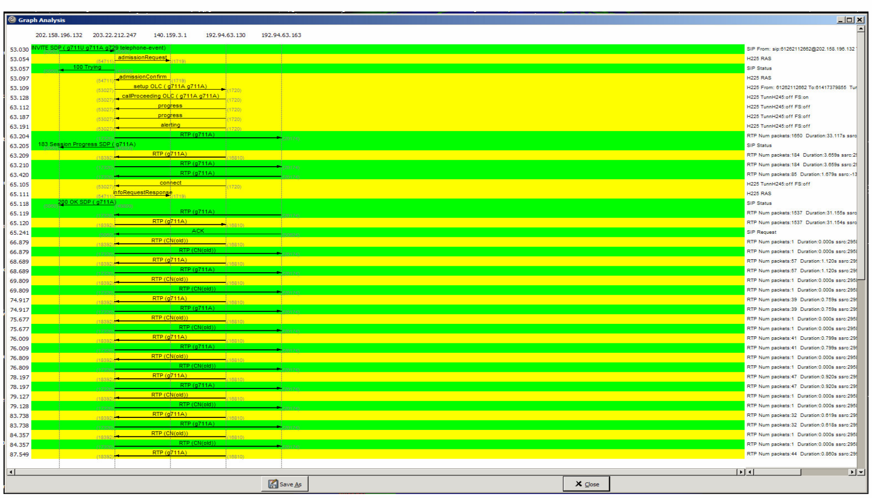
Ethereal: Voip call analysis



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IP to IP Gateway/Gatekeeper – Other uses

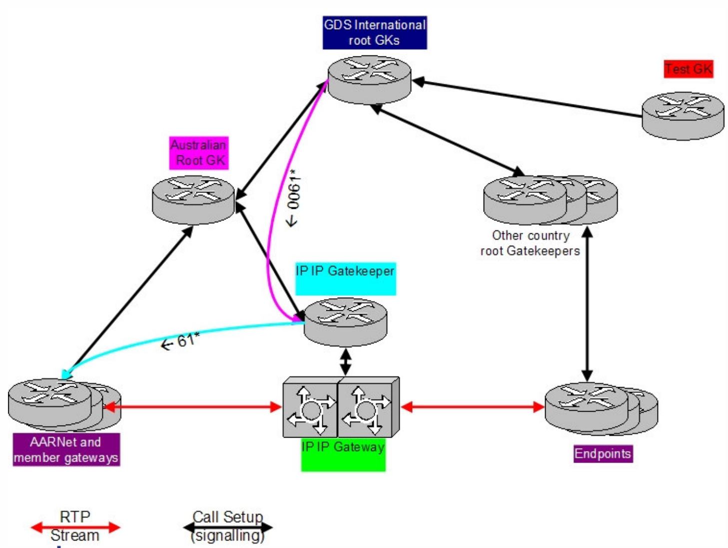
- What are the uses?
  - Peering two H.323 networks
    - Network demarcation (hide internals of network)
    - Dial plan translation
    - CDR billing record collection point

IP to IP Gateway/Gatekeeper – other uses

- What are the uses? (cont)
  - –Gateway to Internet for IP Phones
    - Can act as a “firewall” for IP phones

    Eg. All calls to Internet destinations are made through the IP-IP gateway, so the phone does not need to be visible from the Internet
    - Present a single IP address to the outside world
    - Demarcation point for collecting CDRs from phones

IP-IP International Gateway/Gatekeeper GDS peering configuration



SIP.edu project

- SIP.edu is an Internet2 initiative, the goals of which are to:
- Increase the number of SIP-reachable users within Internet2;
- Promote the convergence of voice and email identities;
- Build a community of Internet2 schools that is experimenting with offering enterprise SIP services.
- By becoming "SIP.edu-enabled", a school makes all of its telephones (and users) reachable via SIP. *

*taken from Sip.edu website, http://mit.edu/sip/sip.edu/

Sip.edu project

- The spot for First Australian University to participate in the SIP.edu project is still up for grabs. Come see me if you want to take the prize..
- All AARNet staff can now be reached at a SIP address, which is the same as their e-mail address.
  - Eg: sip:Ruston.Hutchens@aarnet.edu.au

Reminder…..

Voice over IP Working Group BoF

Today, 17:10 YANDINA ROOM

(not Video Over IP Working group as printed on cards)

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