

# AARNet VoIP update and → peering VoIP

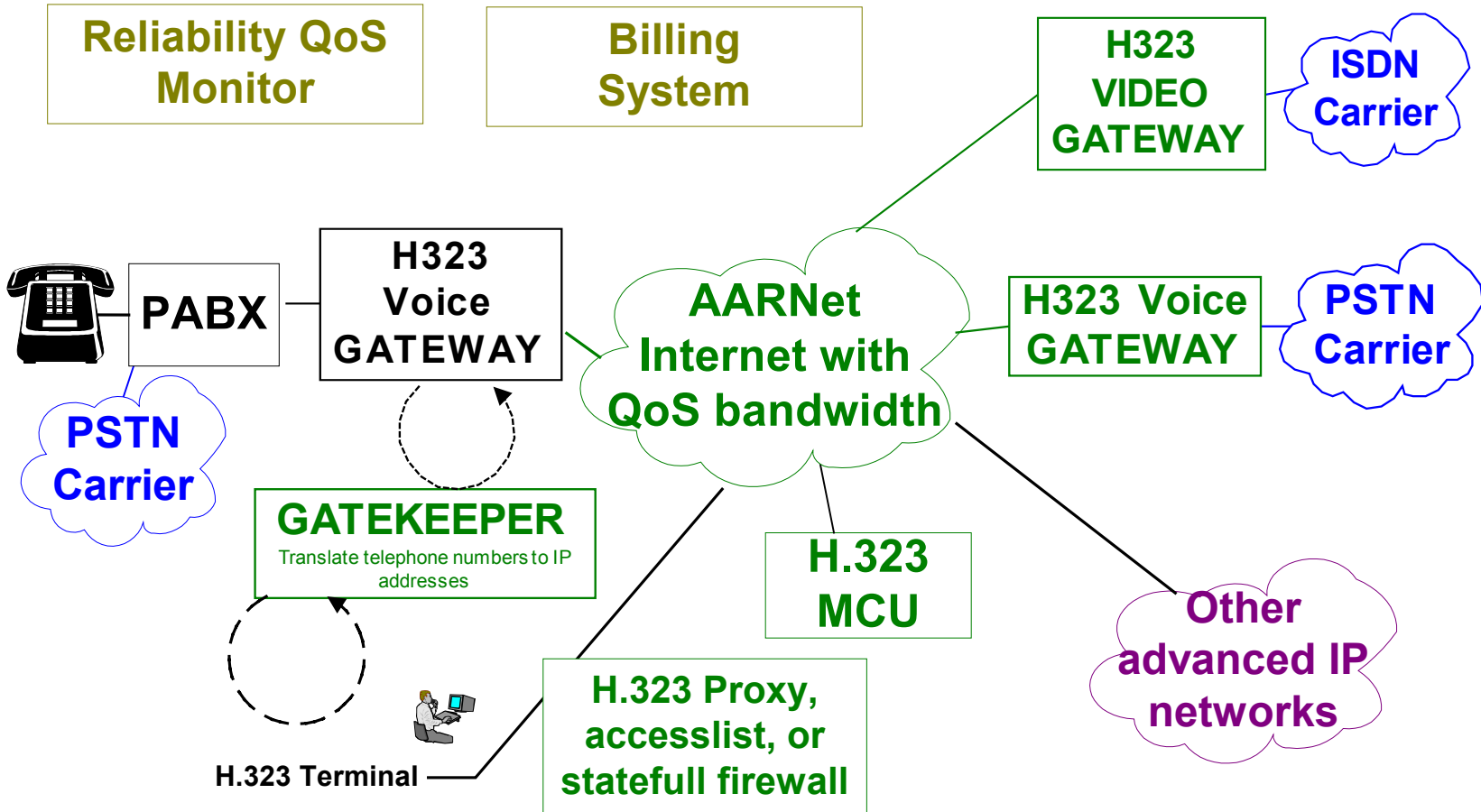
VoIP Summit at APAN Conference and  
Internet 2 Joint Tech's Conference

January 2004

[www.aarnet.edu.au](http://www.aarnet.edu.au)



# → H.323 Architecture AARNet

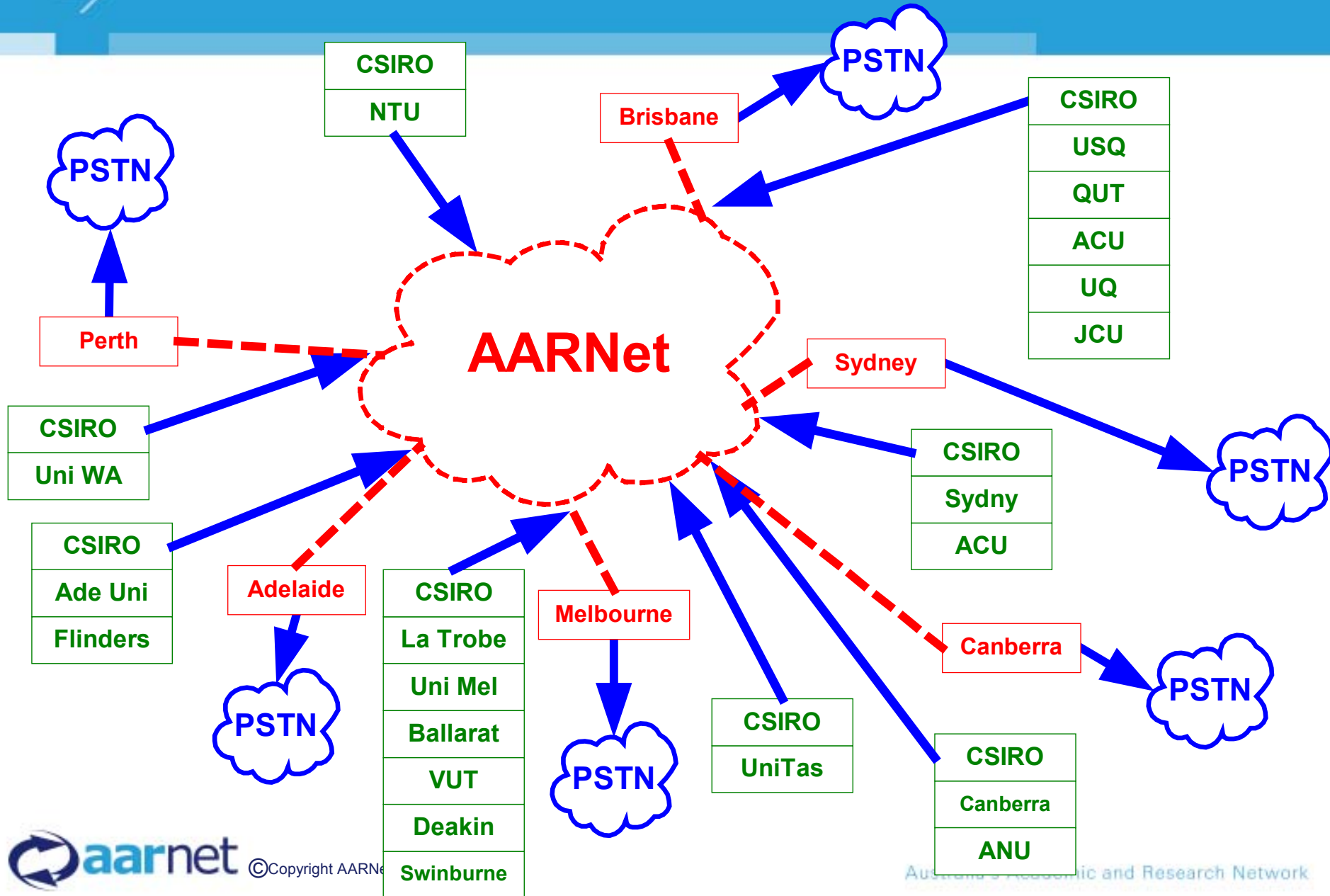


**Based on ITU H.323 standards**

## → AARNet Voice and Video Update

- Started early 1998, operational late 1999.
- Connect 20,000 (average) telephone calls per day.
- Supports commodity telephone calls.
- Supports IP Telephones using H.323.
- Support Video over IP using H.323.
- Supports conferencing for all of the above (in the same conference)
- Seamless connections to PSTN.
- Extremely cheap!
  
- Does not support SIP, although the physical infrastructure can. Major initiative is to enable SIP.
  
- Solved the QoS problem (but not scaleable).
- Solved the billing problem (but not scaleable)
- Solved the Authentication problem (but not scaleable).
  
- <http://www.aarnet.edu.au/engineering/projects/voip/>

# AARNet connections



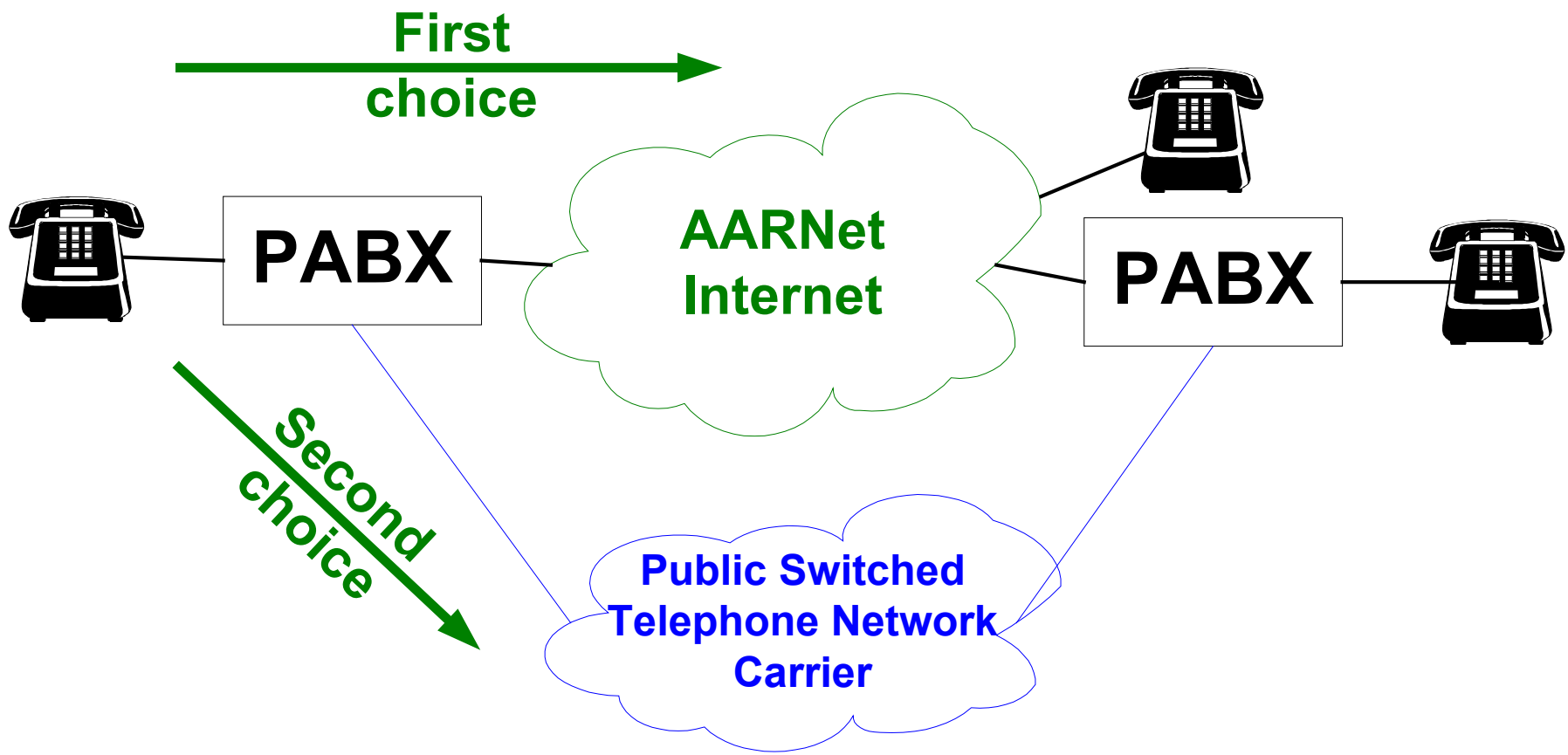
## → QoS implementation

- Selection of good codecs:
  - G.729 over the WAN
  - G.711 on high speed LAN and MAN.
- QoS low latency queues in core WAN routers.
  - IP Precedence 5 for Voice,
  - IP Precedence 4 for Video+Audio.
- QoS Policing today done at edge using accesslists with rate limiting in the core routers. Moving to just having rate limiting at edge to core.
- Plus VoIPMonitor combined with user end point equipment that can alternate route.

## → QoS: VoIPMonitor

- Uses UDP packets (ToS=5) on echo port to measure QoS.
- If there is a failure it tells the gatekeeper where to stop new calls.
- Client Server (threaded application) written in perl. One client at each hub/pop/RNO. Server maintains state and clients log measurements which can be plotted.
- <http://lattice.act.aarnet.net.au/VoIPMonitor>

# “TOLL BYPASS” Business Model



# → Announcement: cheap calls for the research community to about 60% of Australians

- To promote VoIP peering, AARNet will permit VoIP telephone calls to VoIP connected Australian Universities and into the “local call” PSTN zones AARNet services in Australia, being about 61% of the population. Service will be available until it gets abused.
- Contact [Stephen.Kingham@aarnet.edu.au](mailto:Stephen.Kingham@aarnet.edu.au)
  - How to today:  
Point calls using the “International E.164” numbering plan to Australian H.323 Gatekeeper at 203.22.212.247.
  - How to tomorrow:  
USE SIP.



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Australia's Academic  
and Research Network

[www.aarnet.edu.au](http://www.aarnet.edu.au)

## → International Root Gatekeepers for H.323, while we wait for ENUM (or something else).

- Sponsored by ViDe.Net and Radvision
- H.323 based
- Dialplan is 00+E.164 number plan
- Co-ordinated/maintained/supported through a Vide.Net working group (NASM)  
<http://www.vide.net/workgroups/nasm/index.shtml>

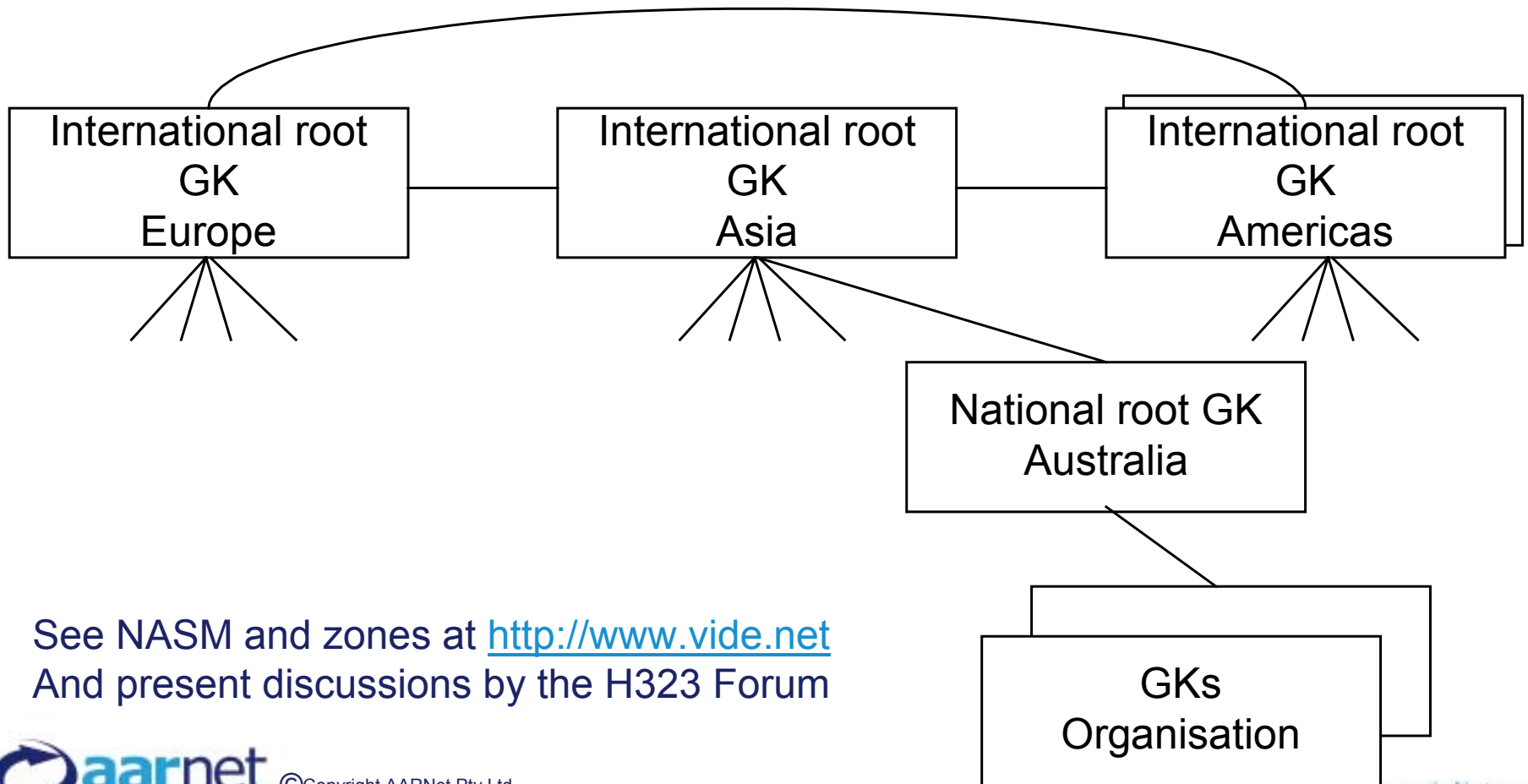
## → ViDe.Net International Root Gatekeepers

### NASM Administrative Domains



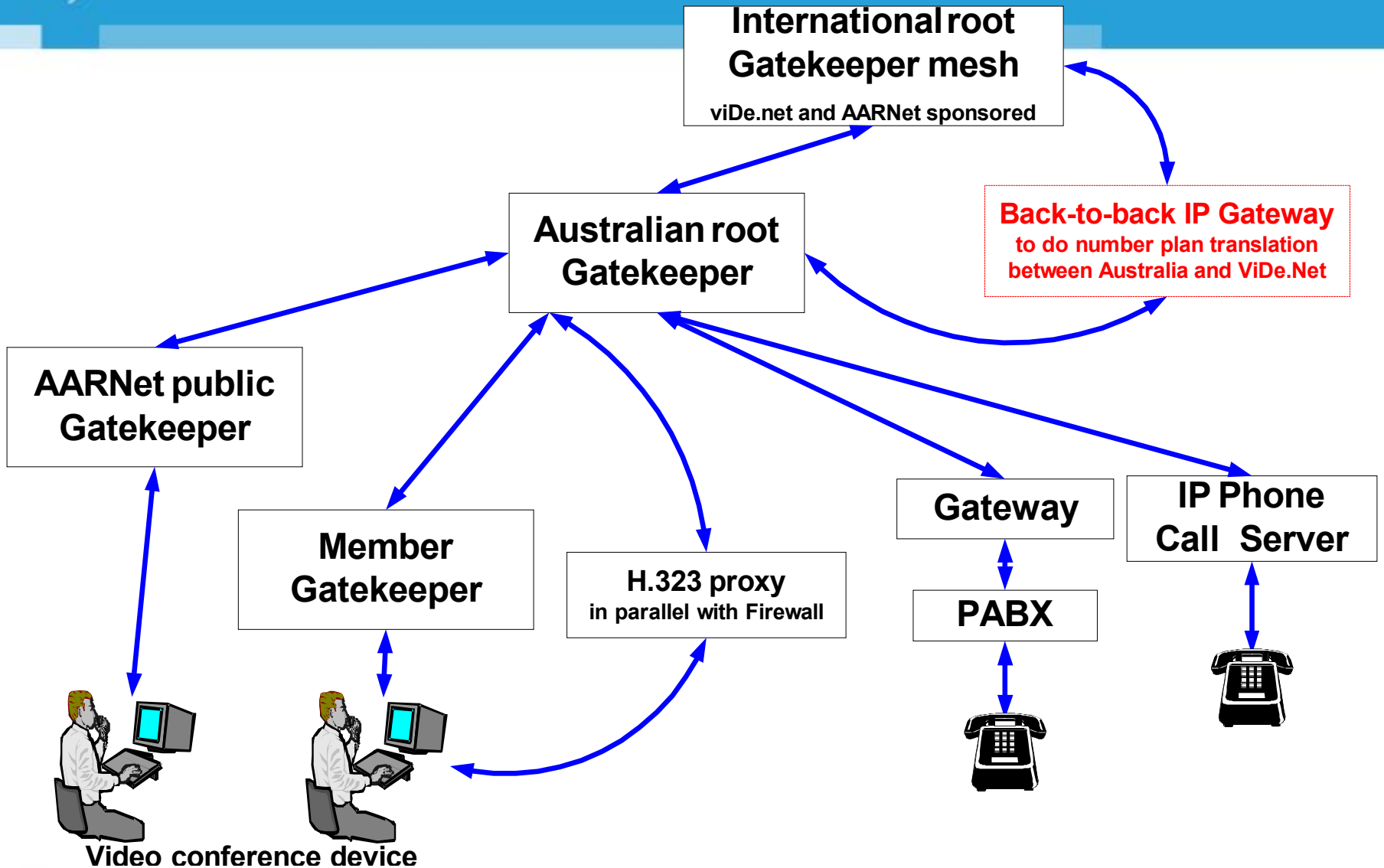
Requests for new or modified address space and general questions are divided among NASM administrative domains, corresponding to large time-zone and geographic clusters to ensure timely and regionally-appropriate responses.

→ While we wait for ENUM we have GK “roots”,  
like the DNS “roots”



See NASM and zones at <http://www.vide.net>  
And present discussions by the H323 Forum

# → H.323 Gatekeeper hierarchy in Australia



## → ENUM

- ENUM is an extension of the existing domain name service by adding a new reverse delegation, eg 5.7.5.3.2.2.2.6.2.1.6.e164.arpa telephone number backwards returns information like sip:Stephen.Kingham@aarnet.edu.au.
- Owners of the delegation is ITU who delegate the Country codes to the Government Agencies in Countries. Consequentially there are huge delays in getting ENUM measured in years (real time, a century in Internet time!). I doubt ENUM will arrive in time before the David Clark “tussle-effect” results in a different solution (ref: Presentation at Joint APAN - Internet2 Joint Techs Conference Jan 2004).
- Is ENUM the answer, or is SIP.edu closer to the best we can achieve?

## → SIP @edu

- See cookbook at <http://voip.internet2.edu/SIP.edu/>
- Requires an entry in the DNS, a SIP Server, and a Gateway to PSTN.
- A person on a SIP endpoint calls a person (not a number) which then rings the person's nominated telephones and SIP end points.

## → Hands on workshop at next APAN meeting

- Objective: Help get people and organisation started
- Over one whole day workshop.
- Activity:
  - Bring a PC with a defined Unix build.
  - build a SER SIP Server
  - Configure some SIP clients
  - Network the servers using DNS, preferably with pre-configured DNS entry in your Organisations DNS.
  - Make calls.
  - Take SIP Server back to home country/organisation.
  - Make calls to other Countries who participate in SIP.edu.

## → New Team in APAN

- Proposal for an APAN Engineering Team, part of the APAN “Network Technology”.
- Email list [h323-sip@apan.net](mailto:h323-sip@apan.net) initially sponsored by Markus.Buchhorn@anu.edu.au
- The group will need a work plan
- What do we need to peer/promote APAN VoIP
  - Dial-plan
  - Who has Equipment, Gateways and or MCUs
  - Who has Gatekeepers (H.323)?
  - Who has SIP?

## → Other possible areas of collaboration

- Routing – TRIP
- Authorisation – who is allowed to make calls to where via my domain
- QoS between Domains
- VoIP interconnect
- Others?