How to provide voice video and collaboration.

"IPTel Working Group"

Questnet 2006.

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Objectives/Audience

Voice, Video and Collaboration

1. WHAT services are there now and what is coming
2. Who uses them and WHY
3. Review of HOW to support them.

The SERVICES

- VoIP which is the same as POTS/PSTN Quality
- IP Telephones to compliment/replace PABX phones
- Room based Video Conferencing

- Roaming IP Telephones.
- Desk top Collaboration.

Each of these are described and discussed in this presentation.

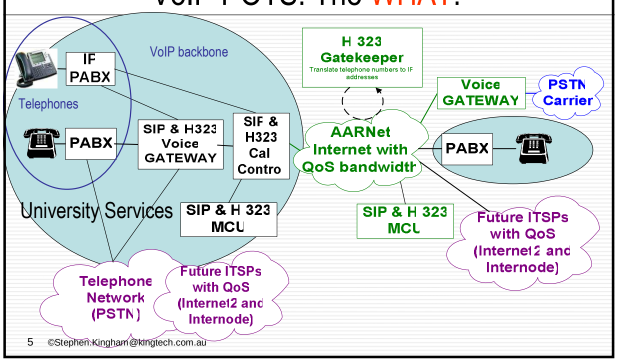
VoIP POTS: The WHAT!

These are technically linked:

- "VoIP POTS" to link distributes campuses.
- "VoIP POTS" to save telephone call charges.
- IP Telephones to replace/supplement PABXs.

And widely used by R&E Sector of Australia and Internationaly.

VoIP POTS: The WHAT!

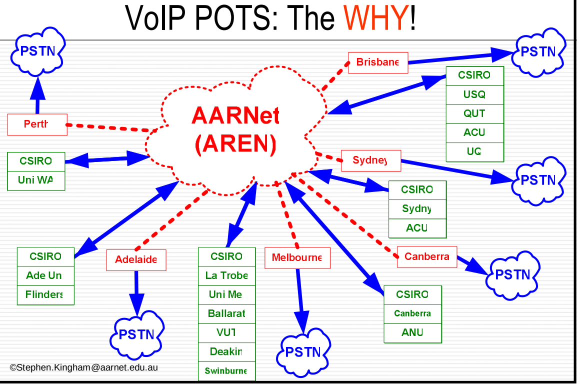


VoIP POTS: The WHY!

- "VoIP POTS" provides a solid base infrastructure for:
  - the PABX replacement IP Telephones, and
  - PABX networking.
- AARNet's VoIP has two functions:
  1. Provide peering between Universities/CSIRO VoIP Nationally and Internationally for POTS quality calls, and
  2. Provide cheap telephone calls.

The technology is well understood and widely deployed.

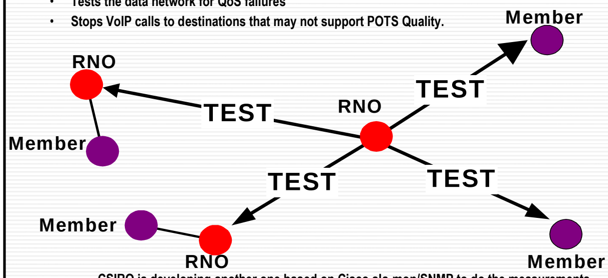
VoIP POTS: The WHY!



VoIP POTS: The HOW!

AARNet uses an application developed by AARNet2/CSIRO to support the POTS quality:

- Tests the data network for QoS failures
- Stops VoIP calls to destinations that may not support POTS Quality.



CSIRO is developing another one based on Cisco sla-mon/SNMP to do the measurements

VoIP POTS: The HOW!

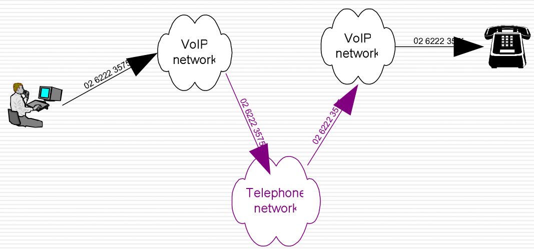
Well understood issues:

- Voice/telephone Team, WAN/LAN Team, and Security Team must be able to work as one Team.
- Power to the IP Telephones
- Safety of Life
- Functionality to support 000, 13 and 1800 numbers.

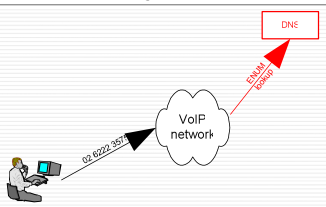
Not so well understood issues (for discussion in Panel and BoF to follow):

- Trust relationships (SPIT and SPIM).
- Scalability of Routing telephone calls between the Universities/CSIRO
- Migration to SIP

ENUM: Routing Telephone numbers using the DNS!



ENUM: Routing Telephone numbers using the DNS!



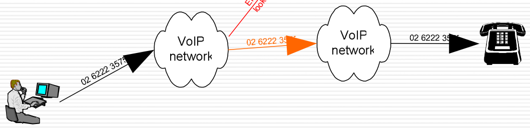
The DNS lookup for a telephone number returns one or more NAPTR records, for example: 61262223575 returns:

sip:+61262223575@aarnet.edu.au

h323:61262223575@203.22.212.245

ENUM: Routing Telephone numbers using the DNS!

There are many private DNS trees to support community based ENUM. e.g. e164.org and freenum.org



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VoIP POTS: The HOW!

Routing

- At present Universities/CSIRO peer to a set of H.323 Gatekeepers in the core. These Gatekeepers are in turn peered to the International H.323 Gatekeepers – thus everyone can everyone else.
  1. Problem: The number ranges are statically set, and thus not very scaleable.
     - Alternative: Use the real ENUM DNS Tree, and a Private ENUM Tree. ENUM can support H.323 and SIP at the same time.
     - Consider enum.org (UQ and Monash have their number ranges there).
  2. Problem: Numbering plans can vary!
     - Stick to one standard as much as possible, and may need to translate (easy with SIP) the rest.

VoIP POTS: The HOW!

Trust relationships:

- Most Universities and CSIRO permit incoming VoIP calls directly through to PABX Telephones and IP-PABX IP Telephones. (eg SIP.edu).
- Some use an access list maintained by AARNet that contains the IP addresses for known H.323 Call Control devices.
  - Problem: Access lists with static IP Addresses is not scaleable.
  - Alternative 1 : Have AARNet set up a special SIP Proxy, only trusted sources can send or will receive VoIP calls from this SIP Proxy.
  - Alternative 2 : Shibbolise SIP Proxies (ref draft RFC).
  - Alternative 3: Use TLS Certificates.

VoIP POTS: The HOW!

Migration to SIP AARNet

- AARNet has a SIP Proxy in the core today which essentially replaces the H.323 Gatekeepers, two engineering development tasks remain:
  1. Get billing records to reconcile the calls that hop onto the PSTN using AARNet VoIP Gateways.
  2. VoIPMonitor integration for the QoS based routing control to guarantee POTS quality
- Then there still remains significant amount of design work.
  - Issue: What about SIP - H.323 conversion??
  - Distinguishing QoS based calls and non-QoS based calls

VoIP POTS: The HOW!

Migration to SIP CSIRO/Universities

- All the PABX and IP Telephone vendors and Carriers are moving to SIP.
- Have a SIP Proxy which faces the outside (the VoIP part).
- Have another SIP Proxy(s) which are used to register users and provide Services (the IP-PABX part).

Non “POTS VoIP”:

- Collaboration is best done today using SIP based technology (eg ECU)
- Consider the SIP.edu initiative.

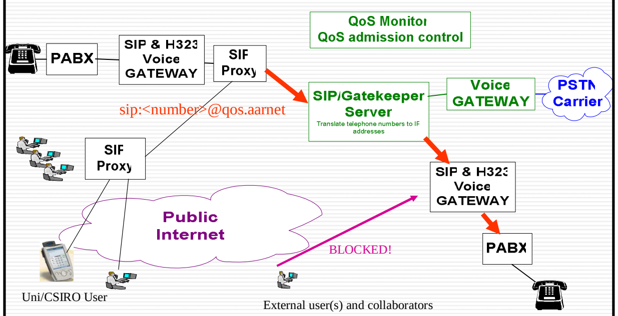
VoIP POTS: The HOW!

Migration to SIP

- IPTEL Working group has suggested the following:

sip:<number>@qos.aarnet	Call must get POTS quality
sip:<number>@qos.local.aarnet	not more than a local call charge
sip:<number>@qos.toll.aarnet	incur long distance change in AU
sip:<number>@qos.free.aarnet	call must be free
sip:<number>@qos.intl.aarnet	call can incur international call rate

VoIP POTS: The HOW!



New Topic:

Roaming IP Telephone!

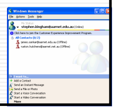
Roaming IP Telephone: The WHAT!

Do not mix these Services, they have different business cases and support requirements, although are technically all linked:

- VoIP to link distributes campuses.
- VoIP to save telephone call charges.
- IP Telephones to replace/supplement PABXs.

AND

- Communication software for Voice, Video, Presence, IM



Roaming IP Telephone: The WHAT!

- SIP is not just Voice and Video, It also has Presence and Instant Messaging.

Roaming IP Telephone : The WHY!

- Mobility, can be reached while you are on the other side of the world (same phone number) – cheaply!
- Cheap telephone calls back to anywhere.

But most of all

- Additional functionality such as Video, Presence and IM.

An example: SKYPE is an island, and it is not SIP!

Has one good lesson: It shows what we need to do for the users!

Lots of negatives:

- Proprietary (secret) protocol.
- Major security accident waiting to happen – as soon as someone reverse engineers the protocol (ref www.voipsa.org/VOIPSEC - VoIP Security).
- User has no control over their bandwidth, eg if they become a Skype Super Node, other people will use your bandwidth.
- No corporate identity, replaced with a skype identity.
- No corporate management.
- Can not integrate with existing infrastructure such as PABX, Video conferencing, Voice Mail, Room based Video, etc.
- It is an island. To call out/in of the island you have to pay $money.

What the customer wants?

Could Universities start losing their customers to 3rd party providers?

Has this already started?

Roaming IP Telephone: The HOW!

- Use SIP based technology.
- Sits on top of VoIP infrastructure.
- No QoS
- This is not your “PABX IP Telephone” so it different Safety of Life issues.

VoIP POTS: The HOW!

Migration to SIP

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Roaming IP Telephone: The HOW!