Future of telephony

QUESTnet2005
Hyatt Regency, Coolum, Sunshine Coast
Queensland, Australia

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Outline and Objectives

- Leading up to today
- Telephone revolution powered by SIP
- What to do now in NRENS and back home

- Demonstration of what is available today

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Telephones BEFORE the 80s see note

- Basic Telephone service
- PABXs generally provided by Carriers, usually on Carrier recommended PABX equipment.
- In Universities it was provided by the “Buildings and Grounds” departments in Universities.

Note: Starting with telephone services based on stored programme controlled TDM based switches.

Telephones in the 80s - deregulation

- Still Basic Telephone service
- Shared structured cabling between LAN and Telephones
- Generally still provided by Carriers. Some private networks using TDM and some tie-lines and voice compression.
- More choice of PABX platform.
- (Tele)Communications Section created by bringing the Voice and Data Communications together as separate Sections under one management group.

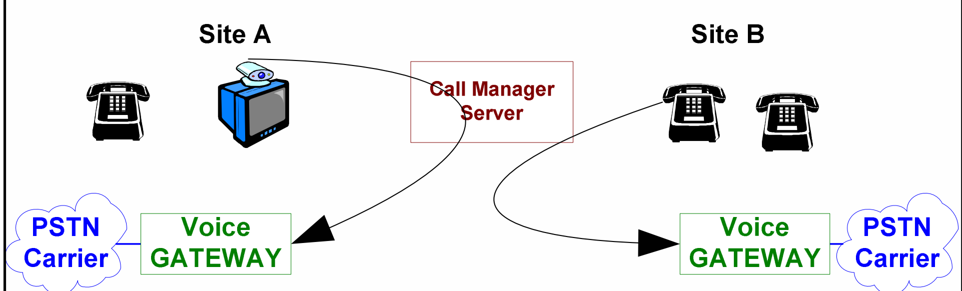
Telephones in 2000-2004 – here comes H.323

- Proprietary IP Telephones deployments:
  - H.323 too hard (although Avaya did it).
  - whole University Campuses (some of the largest Universities in Australia).
  - Some hybrids (IP Telephones with PABX left) and some entirely IP Telephony.
  - IP Telephony based on top of solid VoIP network.
  - Long term better investment and large reductions in adds moves and changes
- VoIP toll bypass based on H.323, tie lines based on VoIP.

- VoIP needs WAN Section to work with Voice Section.
- IP Telephony needs LAN Section to work with Voice Section
- There is a difference between VoIP and IP Telephony

IP Telephones in 2000-2004 – Emergency Services

- Provide POTS phones as power fail/emergency phones, connected to Gateways to the PSTN at each Site.
- Ask how complicated it is to make sure Emergency phone calls are sent to local gateway (easy with new generation PBXs, eg SLIPPER and SER use exception and rules set routing and can use the source ip address?)



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Telephones in 2005+ The impact of SIP and 3rd party Carriers The revolution begins!

- Explosion of SIP UAs and PABXs into the market.
- Many 3rd party providers of sip: accounts.
- Some proprietary solutions (eg Skype) plus some who lock customer in using SIP (eg MSN and Yahoo).
- All the IP Phone and traditional PABX vendors are moving to SIP.
- SIP based PBXs with exceptional capabilities and features, at a fraction of traditional TDM switches.
- Control given back to the user.

- Introduction of the Unix System Administrator (and programmer) skills into the Voice Section.

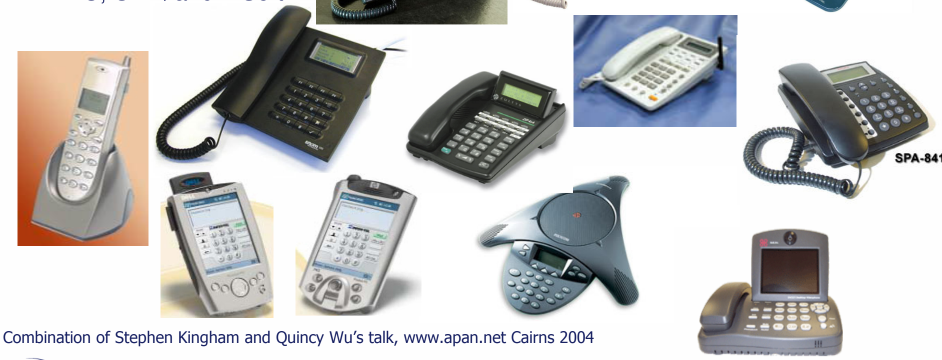
Telephones in 2005+ The impact of SIP

So we have

- VoIP
  - WAN, Gateways, QoS, MCUs, Toll Bypass, different support processes.
- IP
  - LAN, PABX stuff, Emergency Services, built on VoIP, different Business Case to VoIP, different support processes.
- The "roaming IP Telephone" (eduPhone?) - A different animal!
  - Issues…… to be determined.

The impact of SIP
→ Affordable SIP products (NOT H.323)

- Basic SIP IP phones below A$150
- 802.11 phones
- video phones
- Speakerphones
- PDAs with SIP software
- MAC, Unix, and MSoft.



Combination of Stephen Kingham and Quincy Wu's talk, www.apan.net Cairns 2004

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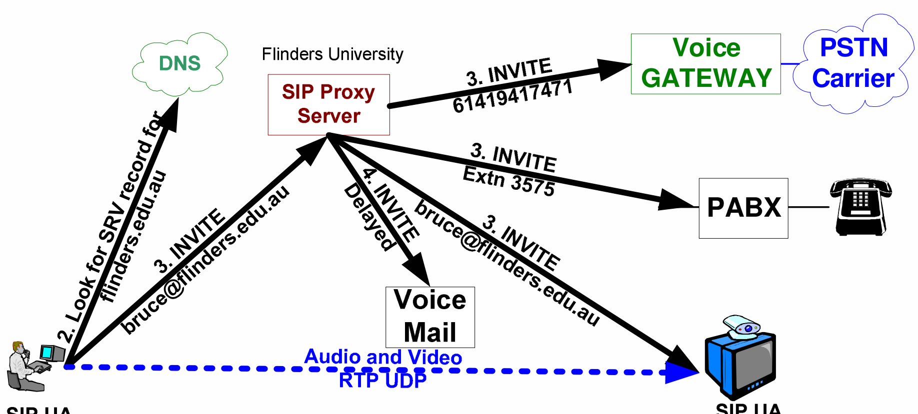
The impact of SIP
→ Something I love about SIP: SIP FORKING (native to SIP)

Never need to forward phones to other phones again!!!!

This is a big change.



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The impact of SIP
SIP based PBXs

Some of these offer exceptional features and capacities

- SIP Express Router (SER) Open Source from www.iptel.org.
  - one config file and mysql
- SIPx (Open Source), causing a lot of discussion
- Asterisk is not really SIP or H.323
  - does some nasty things to the codec negotiations
  - but it is very popular.
  - Great for H323-SIP GW, IVR, and Voice Mail.
  - Many config files.
- Yate (Yet Another Telephone Engine) http://yate.null.ro/pmwiki/
  - Does many things and claims to have a great H.323-SIP gateway.

- There is the start of an explosion of very good quality SIP PBXs.

The impact of SIP
providers of sip: accounts

Provide sip accounts like “hotmail” provides email accounts.

- www.iptel.org (home of Open Source SIP Server SER)
- Free World Dial (fwd) fwd.pulver.com
- www.atp.org (in Australia)
- And many many more, impossible to estimate the number

Providers of closed sip accounts (is this unproductive behaviour?):

- MSN
- Yahoo
- Skype is NOT SIP – and has serious implications (kazaar)!

Could Universities start loosing their customers to 3rd party providers? Has this already started?

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The impact of SIP
Vendors moving to SIP (a sample for discussion)

- NEC
- Avaya
- Cisco new Call Manager is SIP in the core not skinny
- Nortel
- Microsoft

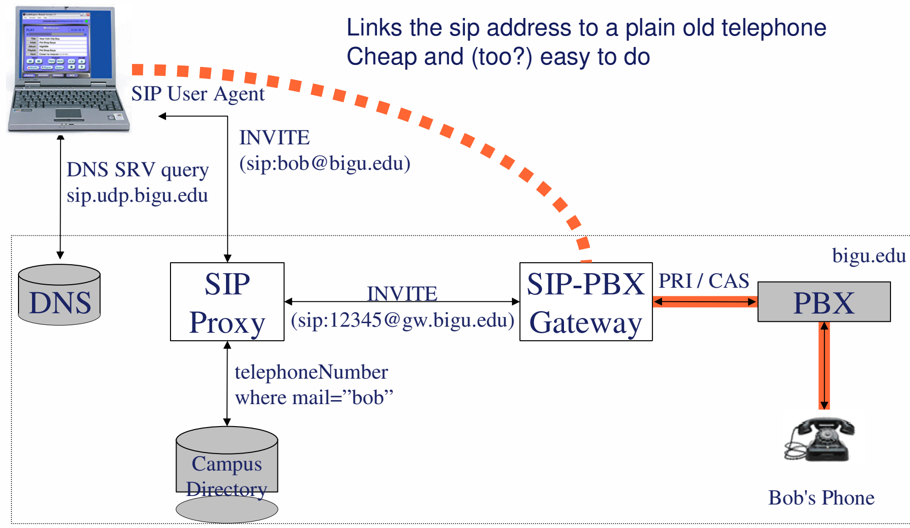
With SIP it is easy to inter-work

The impact of SIP
Here is a possible view of the future (today) a full Voicemail System in Perl (Slipper HelperApp::)

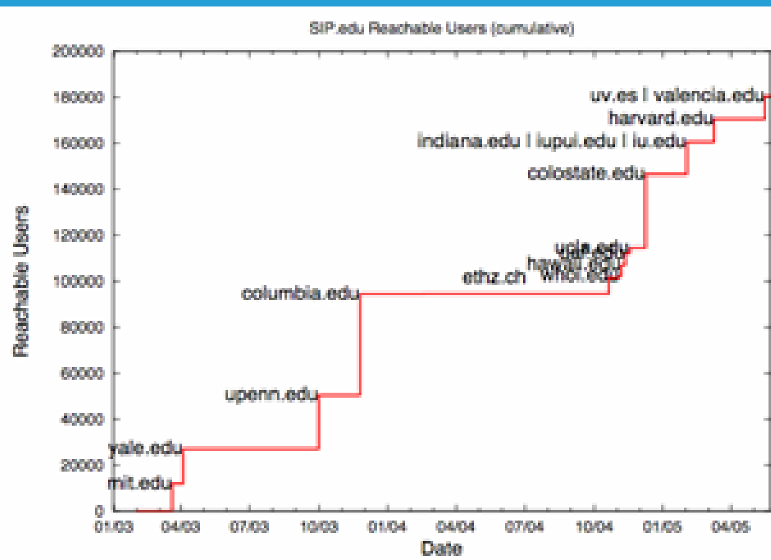
```
#!/usr/bin/perl -w
use strict;
use Slipper::HelperApp;
my $stream = Slipper::HelperApp -> new_stream (shift, shift);
if (! ref $stream) {
      print $stream . "\n";
      exit 0;
}
my $return = $stream -> find_vm_target;
if ($return !~ /^200/) {
      print $return;
      exit 0;
}
$stream -> report_port;
$stream -> play_audio ($stream -> {'VM Greeting'});
$stream -> play_audio ('vm/pling.au');
my ($dtmf, $message) = $stream -> record_audio;
exit 0 if (! defined $message);
$stream -> send_vm ($message);
exit 0;
```

SIP.edu Architecture (Phase 1)

Links the sip address to a plain old telephone
Cheap and (too?) easy to do



SIP.edu Reachable Users



PROPOSAL 1: eduPhone

A precise definition is still needed but here is what we have today:

It is to create a Service similar to fwd and skype.

- Every one can get a SIP account. sip:stephen.kingham@aarnet.edu.au
- Peering to the many 3rd party VoIP Carriers (eg vonage and FWD)

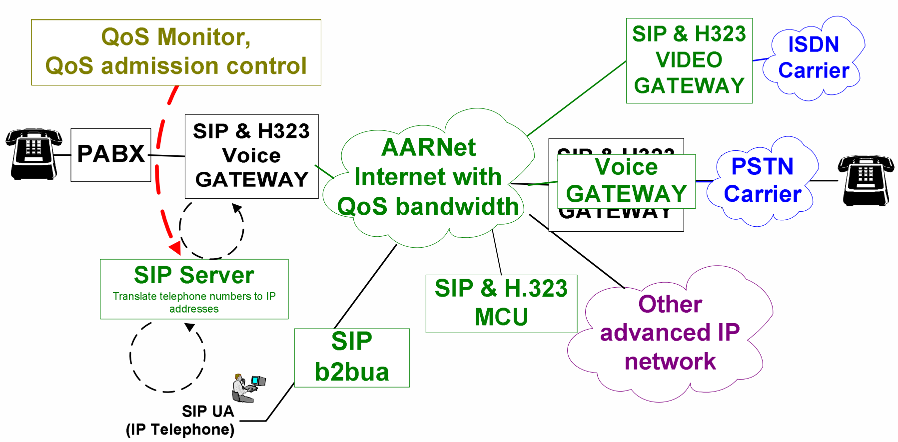
SIP account sits on top of existing Voice, Video, with PSTN connectivity:

Optional extras:

- Hop off to the PSTN
- Hop on from the PSTN
- ENUM number

Members then encourage staff etc to use the University's approved service that can be secured and supported.

where/how to put eduPhone AARNet SIP & H.323 network



The Revolution has started

- SIP enabled IP Telephony and VoIP and VIDEOoIP with Presence and Instant Messaging.
- Mobility, freedom, cheap, flexible.
- Will have a larger Social and Economic impact than http!

- This community needs to work on SIP enablement of their infrastructure and do some leading together with Carriers.
  - Innovation today comes from the worlds ARENs.

    How then do we proceed?

Demo/discussion of a home network

- Lynksys Router/wireless with sipd running
- Analogue SIP gateway
- Hard SIP IP Telephone
- Wireless SIP IP Telephone

References used in this talk

- Internet2 SIP.edu initiative http://voip.internet2.edu/SIP.edu/ take a look at the CookBook.
- http://www.iptel.org/ home of The SIP Express Router (SER)
- www.fwd.pulver.com 3rd party carrier
- SIP Tutorials and Workshops run by AARNet http://www.aarnet.edu.au/events/conferences/2004/sip/

IPTEL and SIP BoF
and
annual AARNet IPTEL Working Group meeting

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AARNet IPTEL Working Group

- Is made up of anyone who joins the email list!

  http://www.aarnet.edu.au/engineering/wgs/iptel/

- Steering Committee is there to:
  - Help make sure there is progress with projects
  - Provide co-ordination
  - Do the administration

  - Rod Ibell (Chair)University of Southern Queensland
  - Steve CoxFlinders University
  - Wil DanielsAustralian Catholic University
  - Ross Dungavel CSIRO
  - Ruston HutchensAARNet Pty Ltd
  - Stephen KinghamAARNet Pty Ltd
  - Richard VanderwaalSwinburne University of Technology

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Agenda

- VoIP Monitor
- eduPhone – New project
- QoS in AARNet3
- Discuss Working Group structure/operation changes
- GNU Gatekeeper
- Charging Model
- Retrofit of SIP (how the two generations run together)
- ENUM project
- ANU AVAYA SIP Project
- Flinders Uni IAGU Slipper SIP Project

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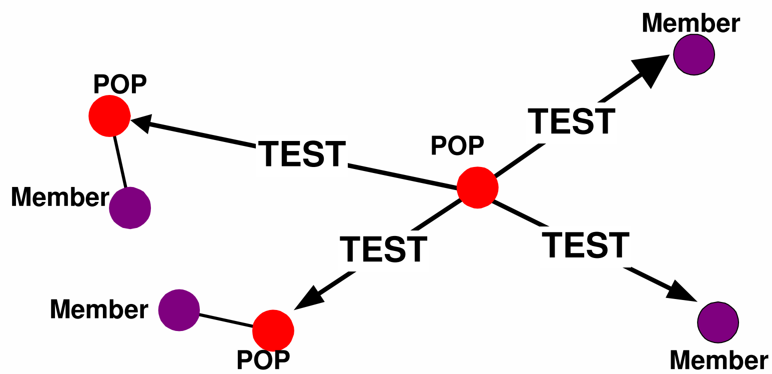
VoIP Monitor version 3 – new project

- Looking at OWAMP for the testing layer
- Active measurement moving to end to end (member edge to member edge).
- Potential to design how Members might run their own



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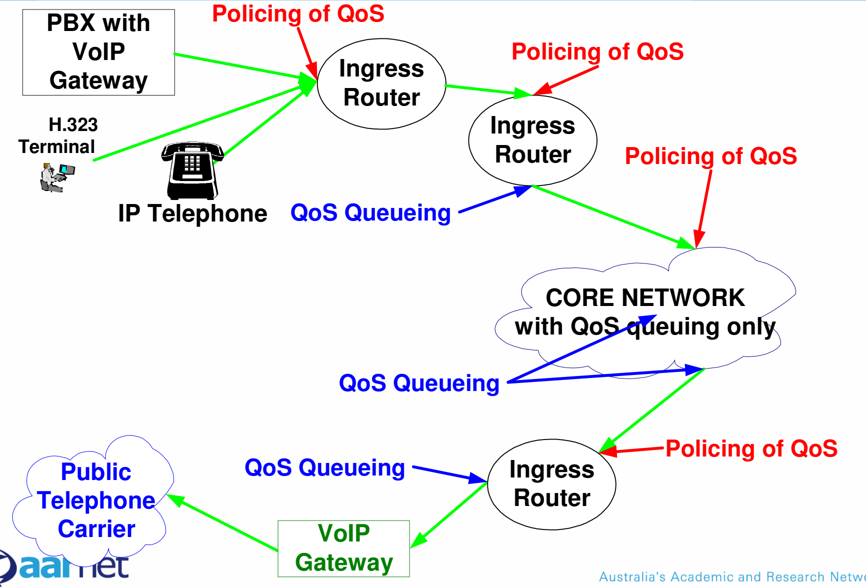
QoS: AARNet3 hands policy control back to Member



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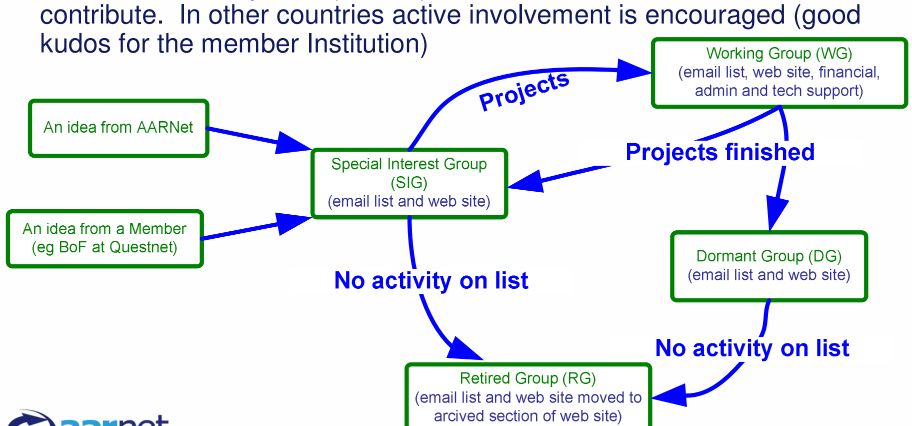
Working Group Changes

- Introduce the concept of stand alone Projects
- Minutes to be distributed and or regular update (newsletter)
- Change the Working Group web sites
- Need to raise the profile (reward) of Members and individuals who contribute. In other countries active involvement is encouraged (good kudos for the member Institution)



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Agenda

- Working Group/AARNet project updates (2 mins each)
  - GNU Gatekeeper
  - Charging Model
    - Moving from 1c/min to a QoS based Billing model is still with the AARNet Billing Model Committee. No change.
  - Retrofit of SIP (how the two generations run together)
    - Gateway do it
    - MCUs do it (just about)
    - Got test/pilot SIP Server in core
    - SIP-H.323 Gateway in the core
    - NOT DONE: Billing of Hopoff and VoIPMonitor integration
  - ENUM project
  - ANU AVAYA SIP Project
  - Flinders Uni IAGU Slipper SIP Project

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