Demystifying VoIP

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aarnet
Australia's Academic and Research Network

Objectives/Audience

1. VoIP – where the "V" is Voice and Video
2. What is IP Telephony, that is only a part of VoIP
3. Roaming IP Telephones are not a PABX replacement
4. IP Telephones include Video, Presence and Instant Messaging.
5. Room based Video fits in where?
6. The Business Case and Support Structures needed
7. An introduction to the standards, the vendors, deployment in AARNet community, and Internationally.

Telephones up to 2005-ish

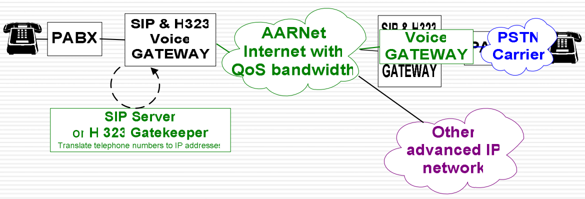
So in summary:

- VoIP
  - WAN, Gateways, QoS, MCUs, Toll Bypass, WAN support processes.
- IP Telephones
  - LAN, PABX stuff, Emergency Services, built on VoIP, different Business Case to VoIP, PABX/LAN support processes.
- Roaming IP Telephone!
  - A different type of IP Telephone!
  - Workstations, Security, and PABX Support processes
- Room based Video
- Desktop Video/collaboration

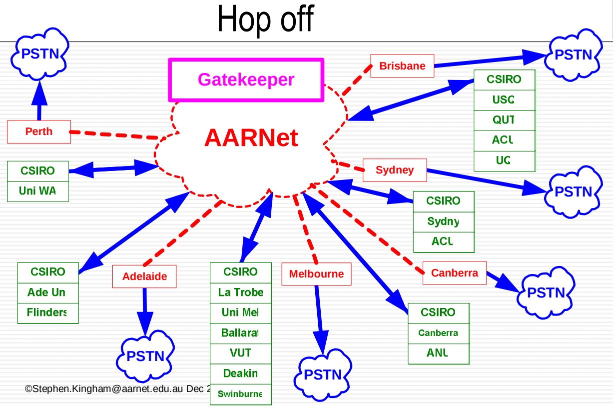
VoIP Standards

1. In 1995 we got the standard H.323. This is a Video Standard from the Carrier world and is based on ISDN.
2. In June 2002 we got SIP from the Internet Standards body (IETF). It uses all the other Internet standards. Is Video, Presence, and Instant Messaging, plus more. Is extreamly simple (read "scary with potential").
3. And we have some proprietary protocols/technology (read "painful").

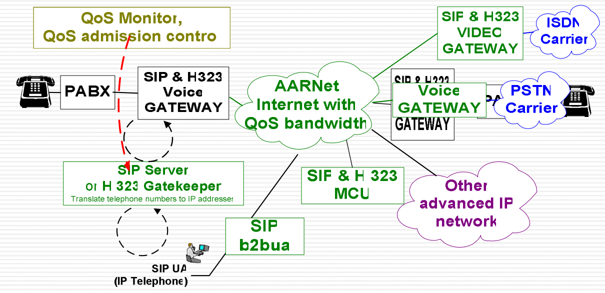
VoIP



Hop off



IP Phone

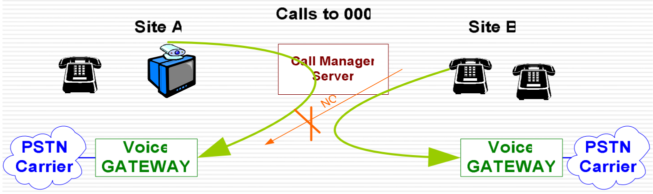


IP Phones: VLAN, POE, QoS

- Put IP phones into a separate VLAN
- IP Phones need power. Either from a power pack, or from the Ethernet switch using POE (Power Over Ethernet).
- Put "power fail" phones in strategic locations, these phones are analogue phones connected to a ATA (Analogue Telephone Adaptor) which is powered with a PABX grade UPS.
- QoS: The LAN must police the use of QoS at the "edge" (as close as possible to the users). Only VLANs with IP Phone (VoIP) can have DSCP = 46 (ToS=5). All other traffic should be marked with DSCP=0.

IP Telephones : Emergency Services

Make sure calls to Emergency Services (000) go to the VoIP Gateway that is at the same site as the IP Telephone.



SIP based PABXs (The SIP Server)

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 – was PingTel), 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.
  - Good for IVR, and Voice Mail.
  - Many config files - dissadvantage.
- Yate (Yet Another Telephone Engine) http://yate.null.ro/pmwiki/
  - Does many things and claims to have a great H.323-SIP gateway.

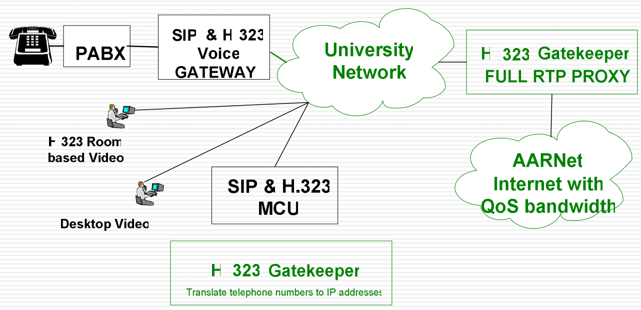
  - There is the start of an explosion of very good quality Open Source SIP PBXs.

Slipper HelperApp: a full Voicemail System in perl:

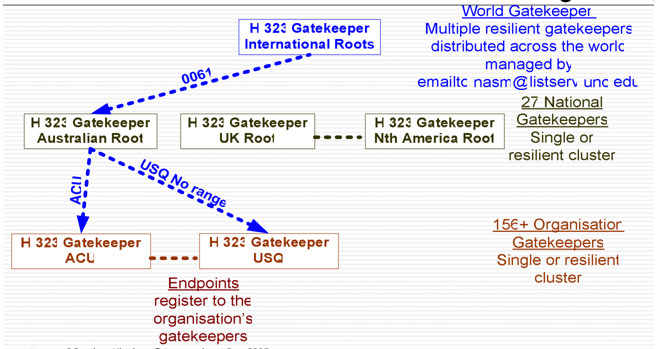
```
#!/usr/bin/perl -w
use strict;
use Slipper::HelperApp;
my $stream = Slipper::HelperApp -> new_stream (shift, shift);
if (! ref $stream) {
      print $stream . "\n";
      exit 0;
}
my $return = $stream -> find_vm_target;
if ($return !~ /^200/) {
      print $return;
      exit 0;
}
$stream -> report_port;
$stream -> play_audio ($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;
```


Room based Video/desk top

Two problems seam to "dog" video conferencing, getting through firewalls and routing



The GDS: International H.323 routing



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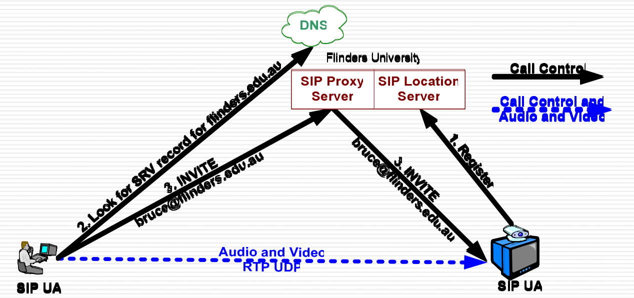
SIP PROXY Server call flow



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SIP Call flow in more detail (Proxy mode)

UA DNS Proxy Server UA

What is SRV record for uni.edu.au

INVITE (with sdp) bruce@uni.edu.au

INVITE (with sdp) bruce@uni.edu.au

100 TRYING

180 RINGING

180 RINGING

200 OK (after call is answered) with sdp

200 OK with sdp

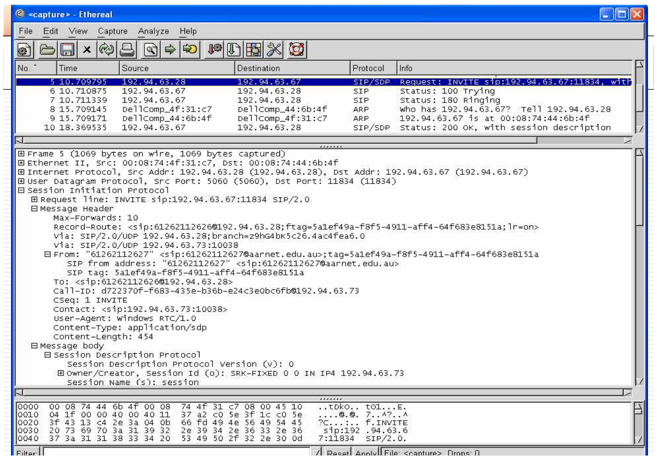
ACK

Media Session, eg Audio and video

Example from NGREP

ngrep -p -q -t -W byline port 5060 sip:

```
interface: eth0 (192.94.63.0/255.255.255.128)
match: sip
U 192.94.63.73:3014 -> 192.94.63.28:5060 INVITE
   sip:61262112626@aarnet.edu.au SIP/2.0..
Via: SIP/2.0/UDP 192.94.63.73:10038..
From: "61262112627" <sip:61262112627@aarnet.edu.au>;tag=484b98c9-2db4 4512-
   95f9-aefc421085b8..
To: <sip:61262112626@aarnet.edu.au>..
Call-ID: 5a144c61-4259-4f59-b121-2931e4011f94@192.94.63.73..
CSeq: 1 INVITE..
Contact: <sip:192.94.63.73:10038>..
User-Agent: Windows RTC/1.0..Content-Type: application/sdp..
Content-Length: 543....
v=0..
o=SRK-FIXED 0 0 IN IP4 192.94.63.73..
s=session..
c=IN IP4 192.94.63.73..
b=CT:1000..t=0 0..
m=audio 55586 RTP/AVP 97 111 112 6 0 8 4 5 3 101..
a=rtpmap:97 red/8000..
etc
```

Response codes used in SIP

- 1xx: Provisional -- request received, continuing to process the request;
- 2xx: Success -- the action was successfully received, understood, and accepted;
- 3xx: Redirection -- further action needs to be taken in order to complete the request;
- 4xx: Client Error -- the request contains bad syntax or cannot be fulfilled at this server;
- 5xx: Server Error -- the server failed to fulfil an apparently valid request;
- 6xx: Global Failure -- the request cannot be fulfilled at any server

Source RFC 3261

SIP and DNS

Specific SRV records added to your DNS for SIP,eg

```
IN A 192.94.63.28
;If we place the SRV record above the next line it fails to load
$ORIGIN aarnet.edu.au.
_sip._udp SRV 0 1 5060
ser.yarralumla.aarnet.edu.au.
_sip._udp SRV 1 1 5060 ser.nsw.aarnet.edu.au.

ser.yarrulumla.aarnet..edu.au.  IN A 192.94.63.28
ser.nsw.aarnet..edu.au.         IN A 138.44.16.90
```


SIP and DNS TEST

- On a unix host use the dig command:
  dig -t SRV _sip._udp.aarnet.edu.au
- You should get a response that has this in it:

```
;; QUESTION SECTION:
;_sip._udp.aarnet.edu.au.    IN  SRV

;; ANSWER SECTION:
_sip._udp.aarnet.edu.au. 333 IN  SRV 1 1 5060
ser.yarralumla.aarnet.edu.au.
```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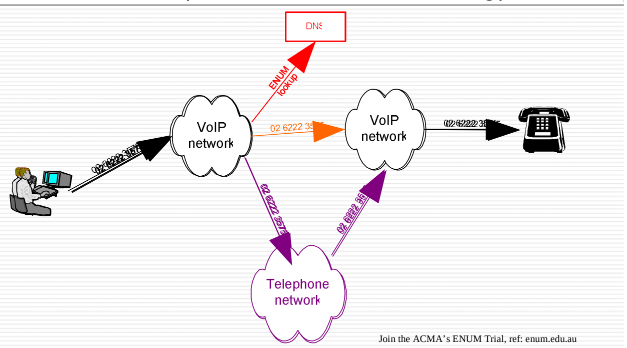
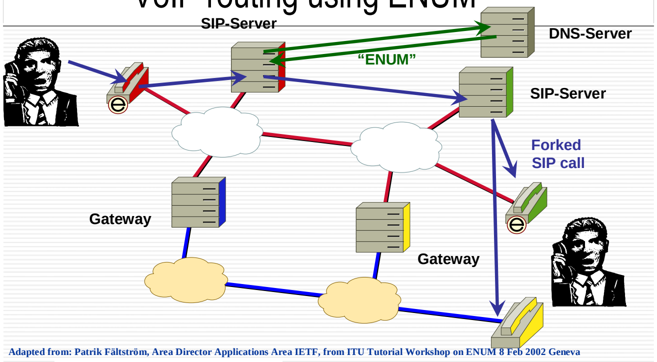
ENUM (SIP and H.323 Routing)



Join the ACMA's ENUM Trial, ref: enum.edu.au

VoIP routing using ENUM



Adapted from: Patrik Fältström, Area Director Applications Area IETF, from ITU Tutorial Workshop on ENUM 8 Feb 2002 Geneva

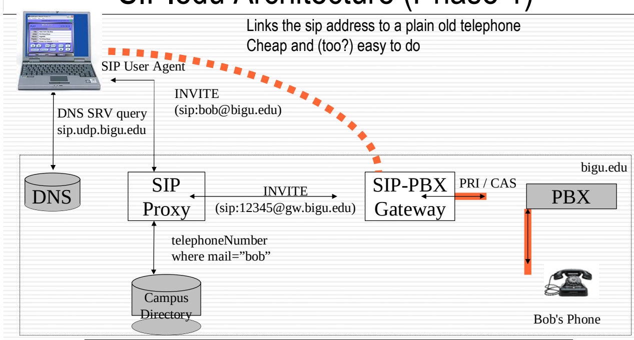
ENUM in a nutshell

- take phone number — +46 86859131
- turn into domain name — 1.3.1.9.5.8.6.8.6.4.e164.arpa.
- ask the DNS
- return list of URI's (NAPTR records) — mailto:paf@cisco.com / sip:paf@cisco.com

Source: Patrik Fältström, Area Director Applications Area IETF, from ITU Tutorial Workshop on ENUM 8 Feb 2002 Geneva

SIP.edu Architecture (Phase 1)

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



SIP Clients

Review slides in presentation yesterday.

This session demonstrates two popular (free) clients.

- Xlite
- Microsoft Messenger

AARNet Voice/Video Services

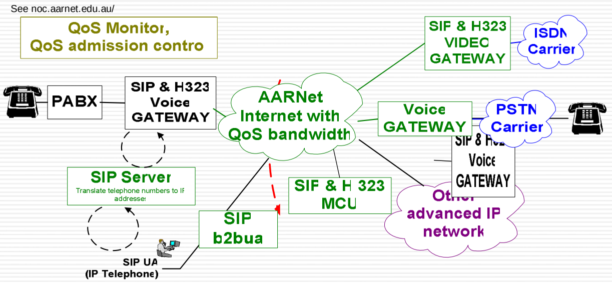
AARNet support Universities and CSIRO.
  Who in turn support their users.

VoIP Toll bypass Service to:
  save money
  support Roaming IP Telephones
Video Service with MCU and ISDN Gateway
  supports Video and Telephone conferencing
  Also web streaming
  Support through a concept of CAVEs.

AARNet SIP & H.323 network

It is the QoS Admission control that enables AARNet connect nearly 10,000 commodity telephone calls per day, started in 1999.

See noc.aarnet.edu.au/



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SIP INVITE with Digest Authentication

UA Proxy Server UA

INVITE fred@uni.edu.au (with out credentials)

407 Proxy Authentication Required

ACK

ask user for a password

INVITE fred@uni.edu.au (with encrypted password)

100 TRYING

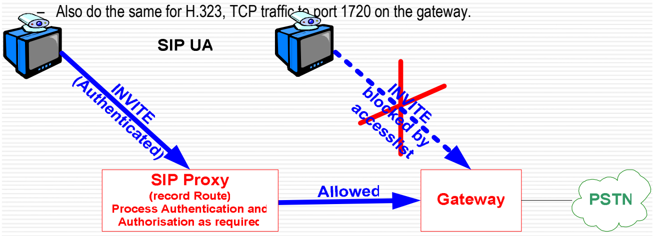
INVITE fred@uni.edu.au (password removed)

Secure SIP

- SIPS, a close cousin of SIP, is a good and low cost means of encryption soon to be widely deployed. It specifies TLS (transport layer security) over TCP and is not subject to bid down attacks. This means a SIPS call will fail rather than complete insecurely.

Protect Gateways from un-authorised use

- Use a Proxy Server in front of your Gateways, turn on Record Route so ALL SIP control is via Proxy.
- Configure gateways so that they only respond to SIP from your SIP Proxy.
  - Filter TCP and UDP traffic to port 5060 on the Gateway.
  - Also do the same for H.323, TCP traffic to port 1720 on the gateway.



What might you do

- Get SIP Server (PABX) going. Use SER (routing and registrations) and optionally Asterisk (for applications like IVR and Voice Mail).
- SIP.edu your University (or just yourself).
- Get involved with Australian ENUM Trial and SIP peering in 2006.
- Get some SIP into your core networks (ie your MCU and PABX)

AARNet: IPTEL Working Group

AARNet: VIDEO Working Group

APAN: SIP-H323 Working Group

Internet2: SIP.edu Working Group

Terena: TF-VVC