AARNet Member VoIP TUTORIAL

How to connect an AARNet Member to the AARNet VoIP Network

15 June 2001, by S.Kingham

Topics covered

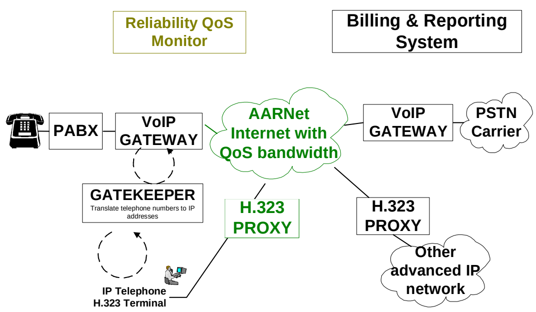
- Building blocks of a H.323 VoIP network
- The PABX
- The H.323 Gateway
- The ISDN protocol
- The H.323 Gatekeeper
- The H.323 Terminal
- The H.323 Proxy
- QoS Reliability
- QoS for guaranteed delay (Glen Turner)
- VoIP vital Statistics
- Billing
- Security
- Futures

Project Background

- AARNet2 - National IP over ATM Based Network where AARNet controls the IP.
- VoIP is one of the AARNet2 Projects
- Voice over ATM- Successful Pilot
- Voice over IP- Successful Pilot
  - ✓Leading Edge/Unproven Technology
  - ✓IP is Ubiquitous in AARNet
  - ✓VoIP, Future Voice Technology?   YES

DR

AATNet VoIP building blocks



Based on ITU H.323 standards

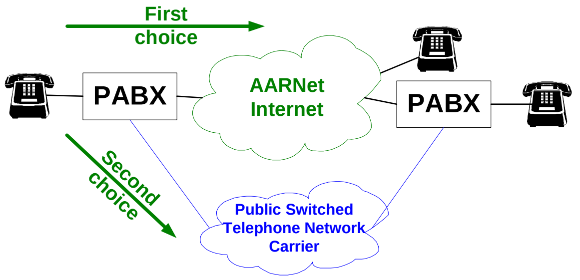
The Other Building Blocks

- Telephony Networking Technicians and Data Networking Technicians MUST WORK TOGETHER as one TEAM!
- Otherwise life will become uncomfortable.

References

- www.aarnet.edu.au
  - **Service**: Brief non-technical service description aimed at Directors etc. A simple description of the service, its costs, etc.
  - **Support**: What the technical folk when the service breaks. Who do you call? What test can you do etc.
  - **Network**: Technical stuff about the network.
  - **R&D**: Where the original project is as it was being developed. History of development etc. Should eventually be historical as the service moves from development to production but right now this is where you find procedures and cook book of how to configure equipment.

"TOLL BYPASS" Business Model



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PABXs

- Keep in mind this is standard telephony interfaces – real "bread and butter" stuff for PABXs – nothing unusual here.
- Interface out of PABX
  - ➢ ETSI ISDN or QSIG.
  - ➢ Do not use CAS
  - ➢ Please note that the Australian ISDN standard TS014 (previously known as TS014 or TPH 014) works but is rapidly becoming unsupported.

PABX sending dialled digits

- Send full national 10 digit numbers (eg 0212345678) or 11 digit international numbers (eg 61212345678).
- Use "en-bloc" sending to the Gateway (it is better than "overlap" in today's networks).
- Caller's telephone number must be a full 9 digit National number, or 11 digit international number. eg extension 5678 on a PABX in Canberra (area code 02) with an indial range 1234xxxx making a call would have:
  - National Calling number of 212345678 or
  - International Calling number of 61212345678.
  - If there is sensitivity then use switch boards number.

PABX Implementation phases

Phases for setting the routing, about one week per phase:

1. Create a route which is accessed by dialling an access code. e.g. if an extension dials 13 the PABX seizes the route to VoIP, then the extension dials the 10 digit. Thus a group of people conscientiously make a decision to use VoIP.
2. For a selected group of extension make it so that their calls automatically go VoIP, with an access code to go via the carrier.
3. Set all users to use VoIP and have automatic alternate routing via the Carrier.

Gateways: key characteristics

- Must provide transparently the same end-to-end signalling and state control used by telephony ISDN and QSIG protocols.
  - ➢ Check the big three call scenarios:
    - ✓ To another ISDN PABX
    - ✓ To POTS
    - ✓ To PABX connected using direct in dial exchange lines (CAS), B chn must connect prior to call being answered so that call progress tones (ring, busy, etc) is heard by caller.
- During IP failure, the Gateway must trigger the PABX to alternate route any new telephone calls. This must be tested, and telephone staff must be able to simulate outage to test this.

Gateways tested/looked at

- Cisco AS5300 with E1
- Cisco 3640 with dual E1, single E1 and BRI
- Cisco 2610 with BRI
- Ericsson (proprietary) – no good for AARNet VoIP
- Nortel enterprise (proprietary) - no good for AARNet VoIP
- 3COM – works but has identified a Cisco/3COM problem. Work is continuing.

- Ask the question – does it work with Microsoft Netmeeting?

"IP Telephone" Business Model

- IP Telephones replace the PABX.
- IP Telephones today are NOT H.323 Terminals, they must have connectivity to a server, somewhere.
- They use the VoIP infrastructure and have the same "toll bypass" advantages.
- Australian Catholic University (ACU) is replacing 80% of their centrex handsets. This is the tip of the installation base.
- Additional infrastructure is required.

IP Telephone infrastructure/disadvantages

- IP Telephones.
- Power, Power packs, from patch panel, from switch and associated UPS issues.
- Still need analogue for fax/modems.
- Conference calls will require DSP farm.
- Switch fabric with QoS, not hubs.
- Firewall (use Proxy for incoming VoIP traffic).
- Policing of QoS (use Proxy for outgoing VoIP traffic).

"IP Telephone" Advantages

- No PABX.
- Simplified building cabling.
- Simplified moves and changes.
- Has "Toll Bypass" advantages.
- Amazing capabilities:
  - True individual roaming (Gatekeeper).
  - Better voice mail and integration to email.
  - Simplified Contact Centre CTI.
  - Address book integration (LDAP/X.500)

Cisco H.323 Gateway debug

- **Working, with 18 calls, use the "show isdn status" command**

```
SydneyGateway# show isdn status
ISDN Serial1:15 interface
        dsl 1, interface ISDN Switchtype = primary-net5
        **** Master side configuration ****
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
    Layer 3 Status:
        17 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 18
        CCB:callid=99EB, sapi=0, ces=0, B-chan=26, calltype=VOICE
        CCB:callid=19FF, sapi=0, ces=0, B-chan=24, calltype=VOICE
                    . . . Lots deleted . . .
        CCB:callid=9AEB, sapi=0, ces=0, B-chan=23, calltype=VOICE
    The Free Channel Mask:  0x80017ABB
    Total Allocated ISDN CCBs = 26
```

- **Not working, eg ISDN cable disconnected.**

```
 Layer 2 Status:
     TEI = 0, Ces = 1, SAPI = 0, State = TEI_ASSIGNED
```


ISDN (and Qsig) Protocol

Use "debug isdn q931"

Extension 02 6276 6223 dials 02 9527 3100 and the call is sent to the Gateway

RX <- **SETUP** callref = 0x2**69C**
    Called Number

TX -> **ALERTING** callref = 0x2**69C**

B channels are now connected through. B end is ringing and A can hear ring tone. 02 9527 3100 answers the call

TX -> **CONNECT** callref = 0xA**69C**
    Progress Ind i = Call not end-to-end ISDN, may have in-band info

RX <- CONNECT_ACK callref = 0xA**69C**

Call is now connected. Users start talking and charging commences.
  02 9527 3100 hangs up.

TX -> **DISCONNECT** callref = 0xA**69C**

Cause i = 0x8090 - Normal call clearing (Note: last 7 bits = 10)

RX <- RELEASE callref = 0xB**69C**
    Cause i = 0x8090 - Normal call clearing

Ref: ETSI standard ETS 300 102-1 for ISDN www.etsi.org

ISDN Protocol - debug

- Use "debug isdn q931" on Cisco Gateways
- Look at disconnect causes, common ones are: Unassigned number, User Busy, No route, congestion. Look at reports in Billing System. Depending on were the cause was generated they can be okay, bad, or network may need some re-dimensioning. (you could refer to the standard, or look at the AARNet web site).

The H.323 Gateway config

See AARNet web site.

http://www.aarnet.edu.au/rd/voip/implementation/howtoconnect.html

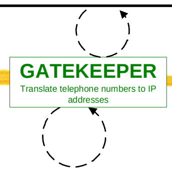
H.323 Gateways – Telephone routing (dial peers)

```
dial-peer voice 10261 pots
 ! Send calls for 02 61xx xxxx to interface E1 1 (local AREA)
 destination-pattern +0261......
 ! In pots peers the matched digits are stripped off, ie 0261.
 direct-inward-dial
 port 1:D
 prefix 61
 ! So the number sent to the E1 interface is 61xxxxxx

dial-peer voice 1073 voip
 ! Send calls for 07 3xxx xxxx to Gateway with ip address 203.22.86.121
 snmp enable peer-trap poor-qov
 destination-pattern +073.......
 fax-rate 14400
 ip precedence 5
 session target ipv4:203.22.86.121
 ! (or session target dns:voice0.uq.aarnet.net.au
 ! or use session target ras to resolve destination using a Gatekeeper).

Dial peer labels have a numbering convention in AARNet.  See
   http://www.aarnet.edu.au/projects/voice/technology/dialPeers.html
```


H.323 Gateway – Telephone routing debug

- Use show "dialplan number 0291234567"
- If there is no dial-peer for a number the Gateway will fail calls with an ISDN clear cause of "unassigned number" (use "debug isdn q931" on Cisco router).
- Use "debug voip ccapi inout" and or "debug voip cc in" to see what is happening in the Cisco gateway.
- Use "debug h225 asn1" to see what is happening on the H.323 VoIP side.
- Use "show gateway" to see if gateway is registered to the gatekeeper.

The Gatekeeper



- Gateways and IP Telephones Register with the Gatekeeper to tell the Gatekeeper where they are.
- Translates Telephone numbers to the remote Gateway or IP Telephone.
- Centralised application of policies and will revolutionise telephone number routing.

H.323 Gatekeeper – config part 1

- Translates Telephone numbers to ip addresses? Actually it looks after Zones.
- Gateways and Terminals register with a Gatekeeper indicating what Zone they look after.
- Gatekeepers can also generate Radius Accounting as well as do user authentication and authorisation.

H.323 Gatekeeper – config part 2

- In Gatekeeper

```
zone local sydney zone.aarnet.edu.au
zone prefix sydney 028.......
zone prefix sydney 029.......
```

- In Gateway

```
interface Loopback1
 h323-gateway voip interface
 h323-gateway voip id sydney ipaddr 152.88.21.21
 h323-gateway voip h323-id voice0.uts@zone.aarnet.edu.au
 h323-gateway voip tech-prefix 1#
```

- The naming convention for zones is going to be important for later.
- Authentication and Authorisation another day!!!!!

H.323 Gatekeeper - debug

- On the Gatekeeper use
  debug ras, and
  debug h225 asn1
- On the Gateway use
  debug cch ras, and
  debug h225 asn1.
- Use "show gatekeeper endpoints" to see what Gateways and Terminals have registered
- Use "show gatekeeper zone status", for a full list of what all the Gateways and Terminals are doing.

Gatekeeper

```
keeper0.anu#show gatekeeper endpoints

CallSignalAddr  Port  RASSignalAddr   Port  Zone Name         Type    F
--------------- ----- --------------- ----- ---------         ----    -
 -
203.21.37.24    1720  203.21.37.24    54574 adelaide          VOIP-GW
    H323-ID: voice0.adelaide@zone.aarnet.edu.au
130.155.1.4     1720  130.155.1.4     54883 eppingcsiro       VOIP-GW
    H323-ID: calamari@zone.csiro.au
130.155.7.1     1720  130.155.7.1     57686 nswcsiro          VOIP-GW
    H323-ID: nswdial@zone.csiro.au
130.155.1.9     1720  130.155.1.9     56515 narrabricsiro     VOIP-GW
    H323-ID: narrabri@zone.csiro.au
130.155.1.11    1720  130.155.1.11    56904 parkescsiro       VOIP-GW
    H323-ID: pks@zone.csiro.au
138.194.5.2     1720  138.194.5.2     54982 melbourne         VOIP-GW
    H323-ID: voice0.unimelb@zone.aarnet.edu.au
203.15.123.98   1720  203.15.123.98   49780 sydney            VOIP-GW
    H323-ID: voice0.sydney@zone.aarnet.edu.au
203.19.110.250  1720  203.19.110.250  50861 perth             VOIP-GW
    H323-ID: voice0.perth@zone.aarnet.edu.au
203.22.212.244  1720  203.22.212.244  54685 canberra          VOIP-GW
    H323-ID: voice0.anu@zone.aarnet.edu.au
LOTS DELETED
```


Gatekeeper Config

```
gatekeeper
 zone local canberra zone.aarnet.edu.au
 zone local melbourne zone.aarnet.edu.au
 zone local adelaide zone.aarnet.edu.au
 zone prefix sydney 029*
 zone prefix geelongcsiro 0352*
 zone prefix tascsiro 0362*
 zone prefix melbourne 038*
 zone prefix vu.edu.au 0396885388*
 zone prefix melbourne 039*
 gw-type-prefix 1#* default-technology
 no use-proxy canberra default inbound-to terminal
 no use-proxy canberra default outbound-from terminal
 no use-proxy melbourne default inbound-to terminal
 no use-proxy melbourne default outbound-from terminal
 no use-proxy viccsiro default inbound-to terminal
 no use-proxy viccsiro default outbound-from terminal
LOTS DELETED
```


Alternative to a Cisco gatekeeper

- Also see http://www.openh323.org/

▪Quality of Service

- Call set up uses TCP, voice transport uses UDP.
- Bandwidth reservation based on IP marked with TOS bits/IP precedence=5, Diff Serv DSCP=40.
- Policing of QoS to be done by RNOs to control Denial of Service attacks – (see URL http://www.aarnet.edu.au/network/design/voip/private/gwusers.txt).
- Using G.729, packet loss can be 30%!!!??????

Gateway - IP (QoS for Bandwidth)

```
interface FastEthernet0
 description LAN connection to RNO router
 ! prioritise ip packets according to the TOS bits
 random
```

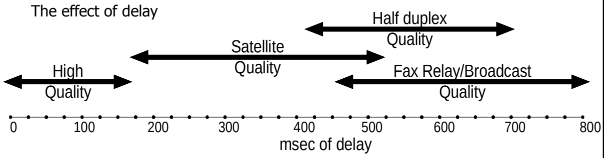
The marking of voice packets is done by the H.323 Gateways and H.323 Proxies using the "ip precedence" command in each "dial-peer". We use TOS bit 5 to differentiate voice traffic from other traffic.

TOS number High to Low Priority	Meaning
7	Critical - network control eg routing updates
6	Flash - Internet control
5	Flash-override eg voice
4	Immediate
3	Internet
2	Network
1	Priority
0	Routine (bulk of the traffic)

Delay

Source: Cisco Conference 1998

The effect of delay



To much delay makes it hard for users, and can result in echo.

Codec delay G.729	25 msec
Queuing delay	6 msec (depends on link speed)
Serialisation delay	8 msec (depends on link speed)
Dejitter delay in end codec	50 msec
Total	89 msec
Propagation delay (one way) to maintain high quality	60 msec one way

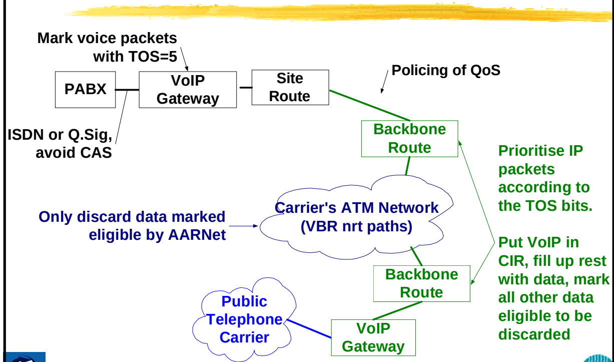
Delay

Source: Cisco Conference 1998

Serialisation

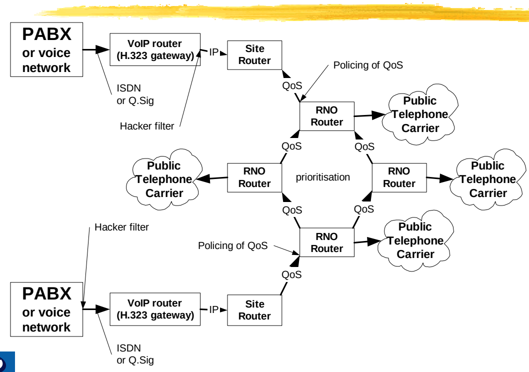
Link Speed	64 Byte Frames	1500 Bytes frames
64 kbps	8 msec	187 msec
128 kbps	4 msec	93 msec
256 kbps	2 msec	46 msec
2,048 kbps	265 usec	5.8 msec

Quality of Service and Policing

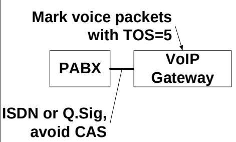


CIR-Committed Information Rate

Interconnection



Quality of Service and Policing



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7	Critical - network control eg routing updates
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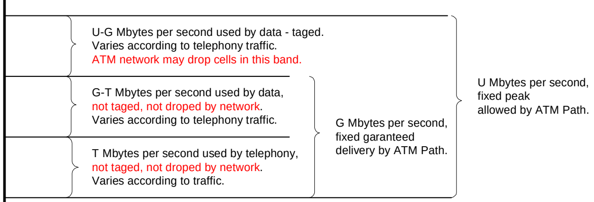
QoS of IP Over ATM BETWEEN RNOs

- Refer to Glen Turner's and Ren Liu's Papers

http://www.aarnet.edu.au/rd/voip/qos/



QoS using Cisco equipment part 1

Use Low Latency Queuing (LLQ):

1. Create an access list to define what to give QoS:

```
ip access-list extended MATCHVOIP&ECHO
 permit ip any any precedence critical
 permit udp any any eq echo
```

2. Create a class-map using the above access list, and limit the bandwidth (eg if for ethernet):

```
policy-map OUTPUTQOSVOIP&ECHOETHERNET
 class MATCHVOIP&ECHO
  priority 768
  class class-default
  fair-queue
```


QoS using Cisco equipment Part 2

3. Apply the Class Map to the outgoing traffic on each interface:

```
Interface serial 0
 service-policy output QOSVOIP&ECHOSERIAL

or

Interface FastEthernet 0
 service-policy output OUTPUTQOSVOIP&ECHOETHERNET

or

interface ATM1/0/0
 service-policy output OUTPUTQOSVOIP&ECHOATM
```

See URL: http://www.aarnet.edu.au/rd/voip/qos/configuration.pdf

QoS using other than Cisco equipment

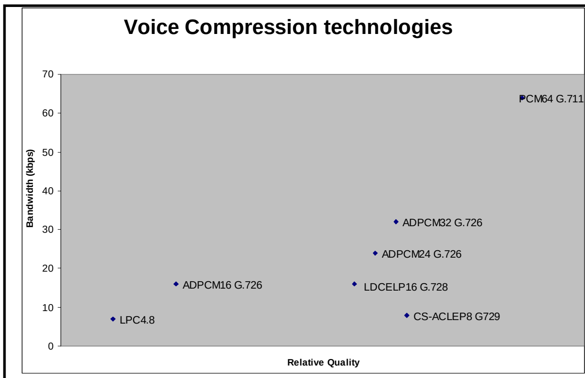
- QUT has Nortel equipment so we should be able to document this equipment.
- Does anyone use anything else?

Policing and stopping VoIP Hackers

- AARNet have to update the two access lists published on the Web
- The RNOs have to implement to Policing otherwise a Member will not get QoS
- Each Member has to implement the Hacker list otherwise a Members calls will not go VoIP, see URL:
  http://www.aarnet.edu.au/network/design/voip/private/gwusers.txt
  This access list also stops Hackers using the UDP echo service, which is needed by the VoIPMonitor discussed later.

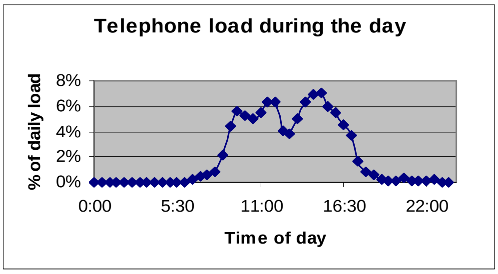
Data bandwidth needed for voice

- AARNet uses is 8kbps ITU G.729 voice compression.
- On the IP side that translates to:
  - 26kbits/sec measured on interface with full time sound.
  - 5kbits/sec average measured by Billing System (average of thousands of telephone calls).
  - 13kbits/sec if RTH compression if used (if link < 2Mbits per second)
- Made up of lots and lots of 60 byte packets.
- Fax is supported using DSP to demodulate that analogue fax signals into data and visa versa.
- Modems and old world ISDN based video conferencing is not supported – PABXs must have an override code to force call via Carrier and not via VoIP.



- Unacceptable is up to about ADPCM16
- Business quality is up to about LDCELP
- Better than LDCELP is Toll quality

Busiest 30 minutes



Based on 54,000 long distance calls by CSIRO sites between major capital cities during October 1998.

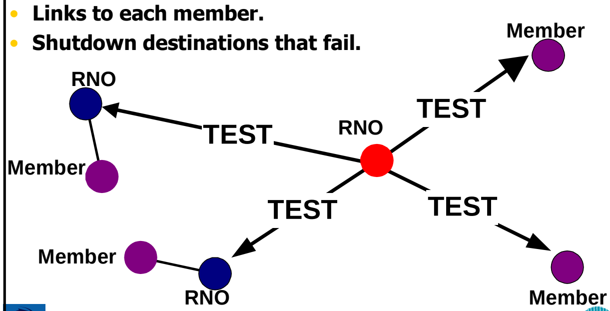
QoS Loss

Reliability QoS Monitor

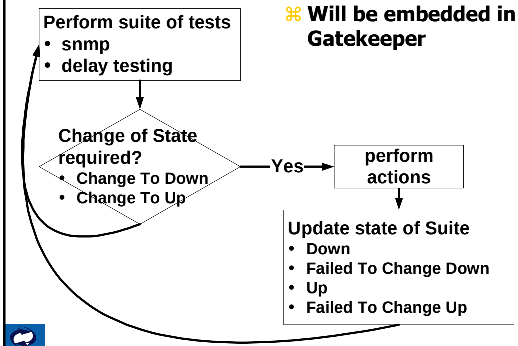
- Guaranteeing Bandwidth for voice is only half the story, packet loss due to errors is often forgotten!!!!!
  - In a test bed with two Gateways using WRED prioritisation connected using CSMA/CD 10baseT running at 60% collision rate, a voice call still works with no noticeable degradation.
- To monitor links that are failing, and to stop VoIP calls when the reliability drops, we have written a new application we have called VoIPMonitor.
- Difficult (or not possible) to put this into the call set up with out advanced networks, more advanced than what exist today!
- AARNet, CSIRO and Australian Universities are in a good position to assist industry in this technology.

VoIPMonitor TOPOLOGY

- **Monitor links to each RNO, and**
- **Links to each member.**
- **Shutdown destinations that fail.**



QoS Reliability: VoIPMonitor



- **Will be embedded in the Gatekeeper**

VoIPMonitor tests used

The VoIPMonitor uses echoping modified by Glen Turner so that packets sent can have the ip precedence set, see URL: http://www.aarnet.edu.au/network/software/

- Cisco routers need to be told to respond to UDP echo. (Some work is still needed to tell the Cisco router to reply with ip precedence 5 set).
- Protection from Denial of Service attacks by the same method of stopping VoIP Hackers http://www.aarnet.edu.au/network/design/voip/private/gwusers.txt

VoIPMonitor Effect

- **Shutdown destinations that fail: all calls in and out of the Gateway will not go VoIP, so in a fully meshed network that is not always desirable.**
- **The occurrence is rare and thus the impact is low.**
- **AARNet needs to implementation a better gatekeeper, one that determines routing using the SOURCE and DESTINATION, rather than just destination.**
- **A www interface is progressing see: http://lattice.act.aarnet.net.au/VoIPMonitor/**

Monitoring QoS: ICPIF

- Is an ITU-T G.113 standard for measuring quality. It replaces the old QDU.
- The Cisco Gateways now send the ICPIF for each call. Actually each Gateway calculates the ICPIF for the receiving VoIP data stream.
- See http://voip.aarnet.net.au/ for the reports.

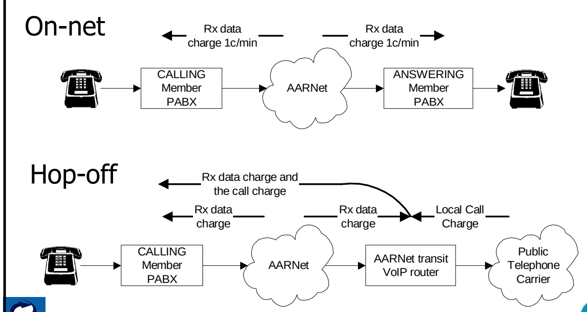
ICPIF

- From ITU-T G.113 standard:

Upper limit	Speech communication quality
5	Very Good
10	Good
20	Adequate
30	Limiting case
45	Exceptional limiting case
55	Customers likely to react strongly

Basis of the Charging Model

- See http://www.aarnet.edu.au/services/voip/



Billing

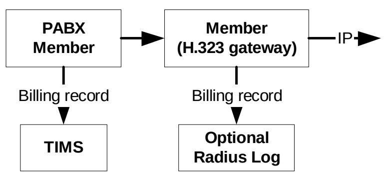
Billing System

- Uses Radius to deliver billing records
- Sophisticated, yet simple billing system written by Bruce Morgan of AARNet.
- See http://voip.aarnet.net.au/

Billing config in Gateway

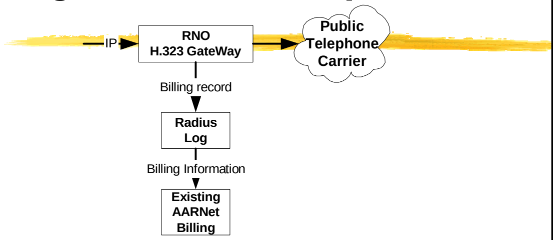
- Gateway will try the first radius server three times.
- If that fails it will try the next, then the next, until it delivers the record or there are no more servers to try.
- Gateway Config, see file.

Billing Infrastructure use existing PABX TIMS if present



- TIMS could use existing charging tables.
  - Little or no changes.

Billing Infrastructure for pure VoIP



- Radius logs:
  - Source and Destination IP
  - Source and Destination phone No.
  - Duration
  - Time

OLD Radius START record

Wed Nov 10 07:53:30 1999
NAS-IP-Address = 130.155.3.2
NAS-Port-Type = Async
User-Name = "0261234567"
Called-Station-Id = "+0291234567"
Calling-Station-Id = "0261234567"
Acct-Status-Type = Start
Service-Type = Login-User
Acct-Session-Id = "42721/07:53:44.100 +1100 Wed Nov 10 1999/riverdial.nsw.csiro.au/418D7733 A2589DA3 0 48791E20/originate/Telephony////"
Acct-Delay-Time = 20
Timestamp = 942180810
Request-Authenticator = Unverified

OLD Radius STOP record

Thu Nov 4 00:24:39 1999
NAS-IP-Address = 130.116.3.2
User-Name = "02621234567"
Called-Station-Id = "031234567"
Calling-Station-Id = "0261234567"
Acct-Status-Type = Stop
Service-Type = Login-User
Acct-Session-Id = "7902/21:19:45.080 +0800 Wed Nov 3 1999/wadial.wa.csiro.au/7DB07DF3 8DE31A83 0 2C37FE4C/answer/Telephony/21:20:13.440 +0800 Wed Nov 3 1999/21:23:32.650 +0800 Wed Nov 3 1999/10/"
Acct-Input-Octets = 263872
Acct-Output-Octets = 68872
Acct-Input-Packets = 8246
Acct-Output-Packets = 1504
Acct-Session-Time = 228
Acct-Delay-Time = 35
Timestamp = 941635479

NEW vsa Radius STOP record

Client-Id = 152.83.5.1
Called-Station-Id = "+xx xxxx xxxx"
Calling-Station-Id = "x xxxx xxxx"
Acct-Status-Type = Stop
Cisco-Gateway-Name = "h323-gw-id=actdial.its.csiro.au"
Cisco-Connection-Id = "h323-conf-id=1978872 5830A8DF 0 E83420C8"
Cisco-Call-Direction = "h323-call-origin=originate"
Cisco-Call-Type = "h323-call-type=Telephony"
Cisco-Setup-Time = "h323-setup-time=12:00:23.794 +1000 Mon Sep 11 2000"
Cisco-Connect-Time = "h323-connect-time=12:00:27.783 +1000 Mon Sep 11 2000"
Cisco-Disconnect-Time = "h323-disconnect-time=12:00:48.403 +1000 Mon Sep 11 2000"
Cisco-Disconnect-Cause = "h323-disconnect-cause=10"
Cisco-Voice-Quality = "h323-voice-quality=0"
Acct-Input-Octets = 21376
Acct-Output-Octets = 11084
Acct-Input-Packets = 668
Acct-Output-Packets = 955
Acct-Session-Time = 21
Timestamp = 968637663

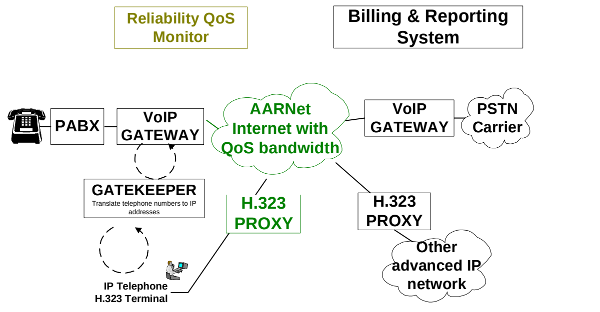
Tools and fault finding

- What does a bad VoIP call sound like: Bits of words missing (QoS), or entire conversation sounds like your talking under water (wrong codecs).
  - ICPIF – Quality measurement based on an ITU standard for each call (Calculated Planning Impairment Factor)

Tools and fault finding

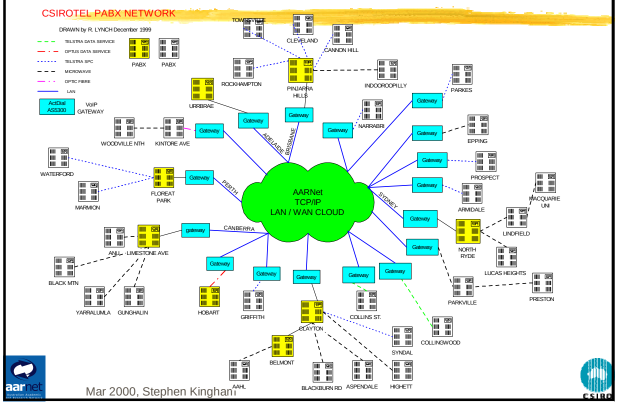
- Use the Reporting/Billing system to find the call and check what the ICPIF was. Fault finding telephony, ie make test calls.
- **Know how to stop new VoIP** calls from being established through a Gateway, ie block tcp port 1720, or on the ISDN interface "isdn service dsl 0 b c 0 state 2"
- Do not allow new calls over low bandwidth backup links.

Review of of the infrastructure



Based on ITU H.323 standards

What CSIRO has now/soon

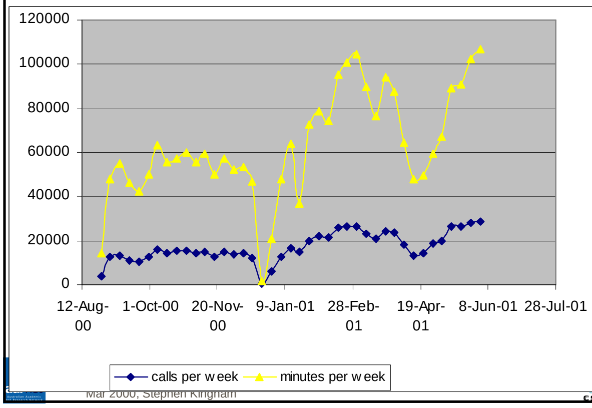


AARNet VoIP Project Status

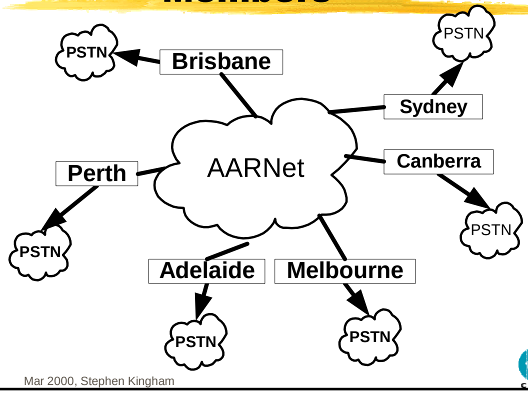
- Switching on average 5,000 calls per working day (12,000 minutes) and expected to explode.
- CSIRO has 12 production gateways and saving 90% compared to 1996 prices.
- There are 5 Universities in production.
- There are 8 Universities with working Gateways and completing pilot phase (or enabling QoS).
- There are many more who are installing Gateways or ordering Gateways
- There are many Universities and CSIRO sites with IP Telephones and growing.

DR

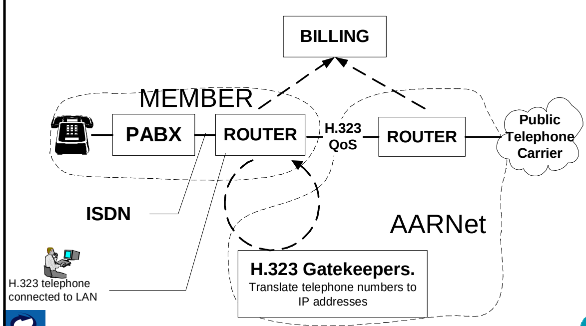
AARNet VoIP Traffic



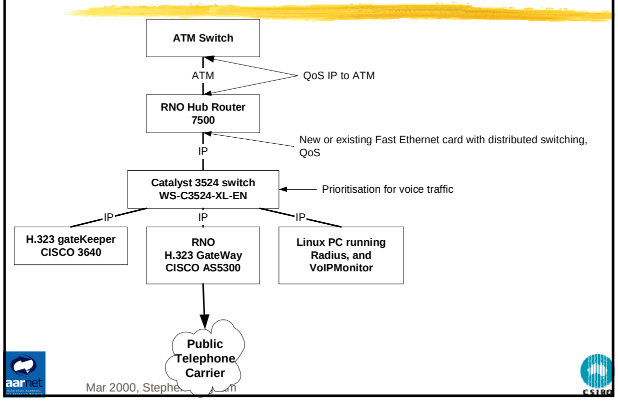
Hop off points for AARNet Members



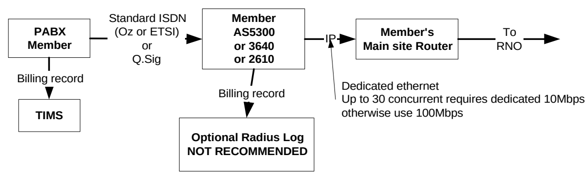
Who supplies what?



Typical Hub



Typical Member



Contact Details and Questions

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