Outline

- VoIP Workshop
- VoIP in AARNet (+)
- Group discussion
- Summary, what’s next?
**Purpose & Program**

- The workshop was initiated and organized by AARNet
- Purpose: to achieve successful collaborative research funding bids (e.g. ARC linkage project)

- Program:
  - Introduction (Chris Hancock, AARNet’s CEO)
  - Beyond the phone call (Dean Economou, SIT CRC)
  - Working groups
  - ARC linkage discussion (Robin Stanton, ANU)

**Participants**

- AARNet (3)
- CSIRO (2)
- SIT CRC (1)

- Cisco (Adam Mulcahy)
- Optus (Narelle Clark, R&D manager)

- Universities: UoNSW (2), Swinburne (1), RMIT (1), Wollongong (2), UoSA (1), Macquarie (1).
**AARNet**

Australia’s Academic and Research Network (AARNet) is a not-for-profit company limited by shares.

- The shareholders are 37 Australian universities (including Swinburne univ.) and the CSIRO.
- Provide services (Internet) for the tertiary education and research sector communities (800e users).
- Wholesale backbone Internet Service Provider

---

**AARNet’s Footprint**

Source: AARNews, issue 1, Sept. 2005

TEIN2: EU sponsored Trans-Eurasian Inf. Net.
VoIP in AARNet


- Infrastructure: IP network, gatekeepers (registration, authentication, addr. translation), gateways (connect to PSTN).

- Also provides network management, billing, quality monitoring.

- In 2002 connect 2400 VoIP devices in 20 institutions.
Voice of the Future

- Talk given by Dean Economou (SIT CRC)
- Main points:
  - Current VoIP focuses on reducing cost, and getting connected to talk.
  - Future: can archive better than “toll” quality? (e.g. fidelity (7kHz); other modalities; spatial sound – echo).
  - Extend personal presence to environmental presence (sound, touch, vision) around the communicating parties.
  - How to measure quality improvements?
Session Initiation Protocol (SIP)

- Purpose: signaling protocol to establish calls between a caller and callee over an IP network.

- Provide mechanisms for the caller to determine current IP address of the callee.

- Calls management (add call, transfer, hold …)

- Single protocol (IETF) for session initiation and management. Does not mandate RTP and codecs.

SIP Cont.

```
SIP client A

SIP Proxy A

SIP Registrar A

redirect response

invite msg.

SIP Registrar B

invite msg.

SIP client B
```
**H.323**

- Complete, suite of protocols (ITU)
- Signalling, reg., admission control, transport, codecs

**E.164 Number and DNS (ENUM)**

**ENUM Cont.**

- Purpose: Use DNS to resolve addresses (RFC 2916)
- No information about QoS, protocol, path, security …
- Private, public, mobile applications are uncoordinated (confusion)
- Update rate, regulatory & policy issues

**Telephone Routing over IP (TRIP)**

- Purpose: exchange information between administrative domains (RFC 3219).
- Similar to BGP but at application level (more complex)
- Can be used to exchange attributes necessary to enforce policies or to select routes based on path or gateway characteristics.
IP Multimedia Subsystem (IMS)

- Purpose: unified architecture to support a wide range of services (spec. for 3G, adopted by ITU, ETSI ..).
- Support multiple application servers: both traditional telephony services (voice) and non-telephony services such as instant messaging, push-to-talk, video streaming, multimedia messaging …
- And can add authentication, presence, gaming …

Possible Objective 2

- Support multi services (voice, video, IM, presence, etc..)
  - ENUM with routing based on E.164
  - Peering between domains with QoS enable protocol (TRIP)
- Generic framework IMS 3GPP (middleware keeping balance between privacy and security)
- Location (Emergency service), IPv6, Security (IPSEC,TLS)
- SPAM, DDoS, Interception for law enforcements
What’s Next

- Forum discussion, working group
- Develop linkage projects
- Set up a working SIP based VoIP using AARNet’s infrastructure.

THANK YOU