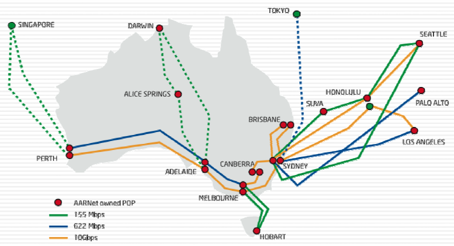
AARNet3 Network



AARNet3

- Provides Internet backbone Services to its Shareholders and to the Research and Education Sector in Australia.
- 10Gb redundant backbone construction
- Provides value add in the Mirror, VoIP Service and VIDEO Service.
- QoS is done on the backbone, new QoS model to be deployed to Members in early 2006.

Personal Communications

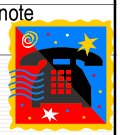
A perspective in history

and

a view of the NOW

Telephones BEFORE the 90s

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 90s - deregulation

- Still Basic Telephone service
- Shared structured cabling between LAN and Telephones
- Generally still provided by Carriers. Some private networks using TDM and some with voice compression (causing problems with FAX).
- 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 1998-2004 – H.323 and VoIP

The VoIP Backbone Is like the WAN

- Still Basic Telephone service
- Business case is around
  - Toll By Pass
  - Supporting IP Telephones and or Video
- Replaced TDM based (much cheaper in support and bandwidth).
- Huge improvement in reliability (especially FAX).
- VoIP needs WAN Section to work with Voice Section.
- Support is internal, not Users!

Telephones in 2001-2004 – here comes IP Telephones

IP Telephones are like the LAN

They replace the PABX.

- Business case is around
  - PABX Replacement
  - Reduce Costs for Adds Moves and Changes
- Technically it is well understood
- Largely proprietary protocols to the handsets.
- Support involves supporting Users.
- IP Telephony needs LAN Section, Voice Section, plus System Administration support.
- There is a difference between VoIP and IP Telephony

Telephones in 2004-2005 roaming IP Telephone

This is the software on the Workstation.

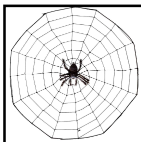
The mobile wireless phone.

- People who travel who want Video, etc.
- People who want to make cheap phone calls.
- Home office.
- Reduced call costs
- Technically it is wellunderstood
  - Access to emergency services is worse/similar to Cellular Mobile Phones.
  - Issues include solving NAT and firewalls.
- Workstation Support to work with Voice Section and Network Security, perhaps also Wireless Network Support.

Telephones up to 2005-ish

So in summary:

- VoIP
- IP Telephones
- Roaming IP Telephone!
- Room based Video
- Desktop Video/collaboration



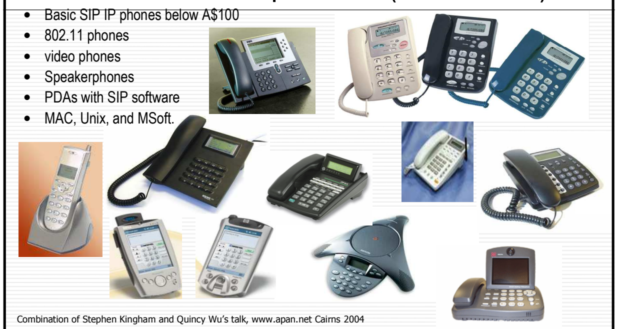
SO WHAT IS COMMING???

SIP:

like HTTP, is a simple standard from the Internet Standard

Affordable SIP products (NOT H.323)

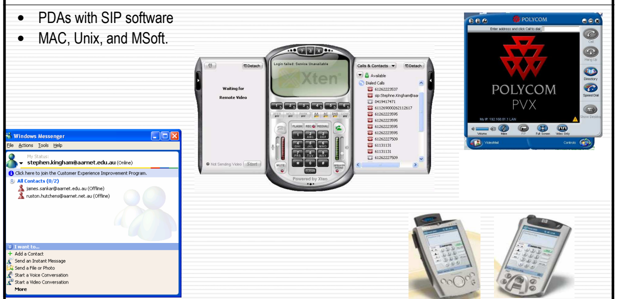
- Basic SIP IP phones below A$100
- 802.11 phones
- video phones
- Speakerphones
- PDAs with SIP software
- MAC, Unix, and MSoft.



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

Also SIP Clients

- PDAs with SIP software
- MAC, Unix, and MSoft.



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

SIP based PABXs (The SIP Server)

- SIP is so easy to develop in.
- Many quality Open Source SIP PABXs.
- Some of the VoIP Carriers use these Open Source Products!
- They include Call Routing, Forwarding, IVR, and Voice Mail.
- All the PABX Vendors are moving to SIP based technology.
- All the Carriers are deploying their VoIP and IP Telephone Services using SIP Technology.
- With SIP it is easy to mix and match products.
- SIP is really easy to support.

SIP based VoIP Carriers (too many)

ENGIN
AAPT
Internode/Agyle
ATP

ITouchTone
AOL
AT&T
BroadVoice
Broadvox Direct
Dialpad
Galaxy Voice
Global Village
G02Call

G02Call
iConnectHere
InPhonex
Lingo
Mutualphone
MyPhoneCompany
Net2Phone
Nikotel
NuFone
Packet8
QuantumVoice
SimpleTelecom
SIPphone
SIPphone
Skype (not SIP)

StanaPhone
SunRocket
TeleSIP
TellAX
TerraCall
USA Datanet
VoiceGlo
VoicePlus
VoiceWing (Verizon)
VoipJet
Vonage
VoxFlow
WebPhone
Yahoo
ZipGlobal
IIC (old ozemail)

There are some key questions to ask.

VoIP Carriers that provide 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 more

Providers of closed sip accounts:

- MSN
- Skype (not SIP)
- Most do not permit calls to or from other VoIP provides.

SIP based VoIP Carriers (a small sample)

- ENGIN, buy a black box from Dick Smith (no QoS).
  - 10c (untimed) to any Australian number, 29c/min to mobiles, free to another engin user, 3.5c/min to key international destinations.
- Internode combined with the Internode ADSL (has QoS).
  - 18c (untimed) to any Australian number, 30c/min to mobiles, free to another internode number, 15c/min to key international destinations.
- Free World Dial (no QoS), provides a SIP account
  - Call other SIP addresses.
  - Call other VoIP Networks using an access code.
- AARNet (with QoS)
  - 6c plus 1c per minute to 90% of Australians, 25c/min to Mobiles.
  - Only available to AARNet Member Organisations.

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.
- Loose corporate identity, replaced with a skype identity.
- 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.

€ 0.017 (about A$0.027c) per minute to Australian Numbers is more expensive than AARNet and Engin.

What the customer wants?

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

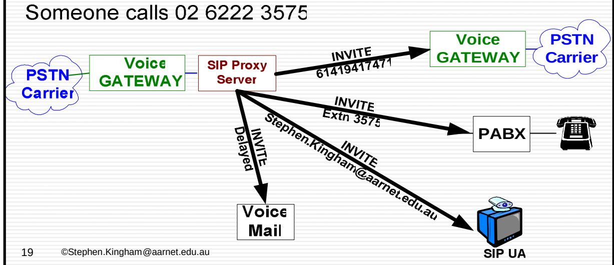
Has this already started?

SIP FORKING (native to SIP)

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

This is a big mindset change for the user.

Someone calls 02 6222 3575



All the Vendors moving to SIP

- NEC
- Avaya
- Cisco new Call Manager is SIP in the core not skinny.
- Nortel
- Microsoft (PABX functionality soon)
- An Australian Product called Slipper by IAGU.

Both Avaya and Cisco integrate the PC with the IP Telephone to make a user friendly Video phone.

With SIP it is easy to inter-work.
Voice mail and IVRs are very easy.

SIP Addressing, will it be the future?

A. G. Bell did **not** say:

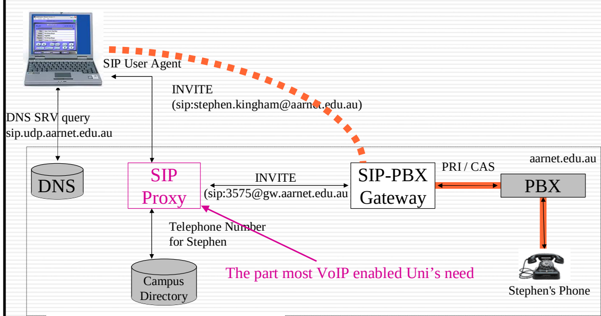
"+61-2-6222 3575, come here. I need you!"

© Ben Teitelbaum @ Internet2

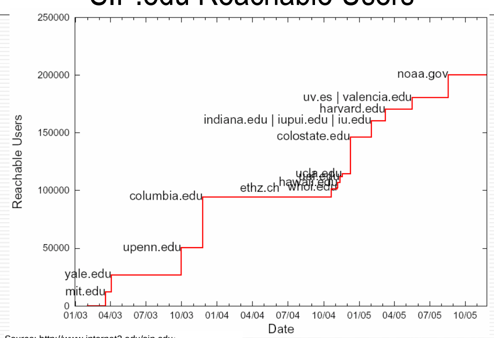
sip:Stephen.Kingham@aarnet.edu.au, or srk@aarnet.edu.au
Within the next year you will see this on the bottom of more email footers and on business cards.

But we will also want to support the 12 button user device (ie key pad)

SIP.edu Architecture: An achievable goal



SIP.edu Reachable Users

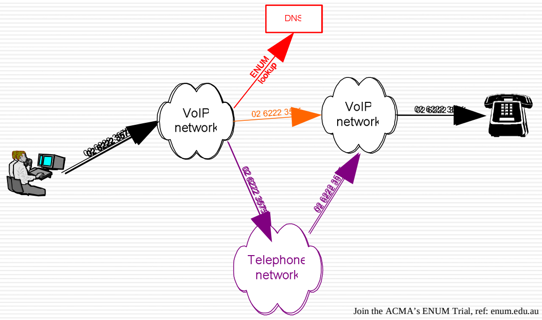


Routing Telephone numbers!

- The WWW and email function because of the Domain Name Service.
  - The DNS turns human addresses into Internet addresses,
  - the DNS on it's own is very uninteresting or useful!
- The ENUM standard teaches the DNS about Telephone numbers!
  - VoIP users can discover that they can make VoIP calls to a number without routing it first to the PSTN!
  - Traditional Carriers around the world do not like ENUM.

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

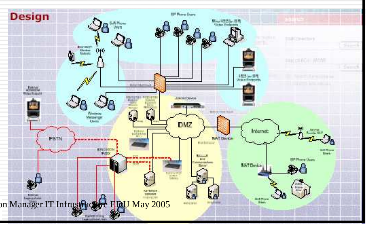
Routing Telephone numbers using the DNS!



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

Case Study from Edith Cowan University

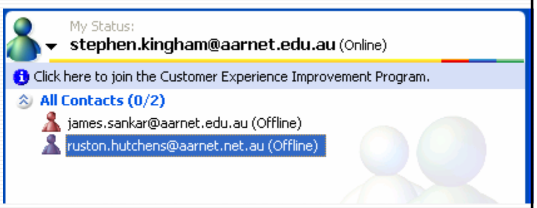
- SIP Enabled their core.
- SIP integrated Voice, PABX, Room based Video, Desktop Video, mobile SIP phones on campus, Instant Messaging and Presence.
- Unexpected demand was the Presence and Instant Messaging.



Source: APAN 2005 and Questnet 2005, Steve Johnson Manager IT Infrastructure EDU May 2005

Presence and Instant Messaging

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



The Revolution has started

- Control given back to the user. No more forwarding calls.
- Presence and instant Messaging.
- Introduction of the Unix System Administrator (and programmer) skills into the Voice Section.
- Lots of hype and confusion in the market place. – watch out for destructive events (skype, and SPIT).
- The telephone will be unrecognisable.

Look forward to lots of sipping