SIP Tutorial

VoIP Workshop
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Australia's Academic and Research Network

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Outline and Objectives

- What is SIP
- Building blocks of a SIP Network
- SIP Routing
- Overview of SIP control protocols
- Overview of SIP media negotiation; SDP
- Fault finding SIP control protocols
- SIP Network in AARNet community
- Why SIP will replace H.323

SIP Standards

- IETF (Internet Standard) RFC 3261 (replaced RFC 2543)
- Because SIP is an “internet” standard it leverages and is leveraged by other internet services:
  - Instant Messaging RFC 3428
  - Presence (The SIMPLE Protocol)
- Uses SDP, Session Description Protocol RFC 2327, ie what codecs to use.
- And because SIP is an IETF standard it is designed to fit in with all the other Internet standards. Eg ENUM.

SIP is PBX/Centrex ready

centrex-style features

call waiting/multiple calls	RFC 3261
hold	RFC 3264
transfer	RFC 3515/Replaces
conference	RFC 3261/callee caps
message waiting	message summary package
call forward	RFC 3261
call park	RFC 3515/Replaces
call pickup	Replaces
do not disturb	RFC 3261
call blast	RFC 3261

boss/admin features

simultaneous ringing (forking)	RFC 3261
basic shared lines	dialog/reg. package
barge-in	Join
"Take"	Replaces
Shared-line "privacy"	dialog package
divert to admin	RFC 3261
intercom	URI convention
auto attendant	RFC 3261/2833
attendant console	dialog package
night service	RFC 3261

attendant features

from Rohan Mahy's VON Fall 2003 talk

What is SIP: Use perspective

- Heaps simpler than H.323
- It is easier to support than H.323
- Lots more products than H.323
- Cheaper than H.323
- Does more than H.323
- Has digest authentication (encrypted shared key for users)
- In practice SIP has more features than H.323.
- All the Video Manufacturers are moving to SIP!
- All the PABX manufacturers are moving to SIP!
- Find a H.323 client for Mac or Unix – no you can not!
- But there are SIP clients for Mac, Unix, PDAs, Microsoft messenger is a Video capable SIP client (support G.722.1 and H.263), GPRS, G3, ….!

Lesson: do not do anything with Voice, Video, Telephony, Instant Messaging or Presence unless it uses SIP.

SIP Forking: Introduction

- SIP natively does forking: Make several phones and UAs ring all at the same time. The call is connected to the UA that answers the call.

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

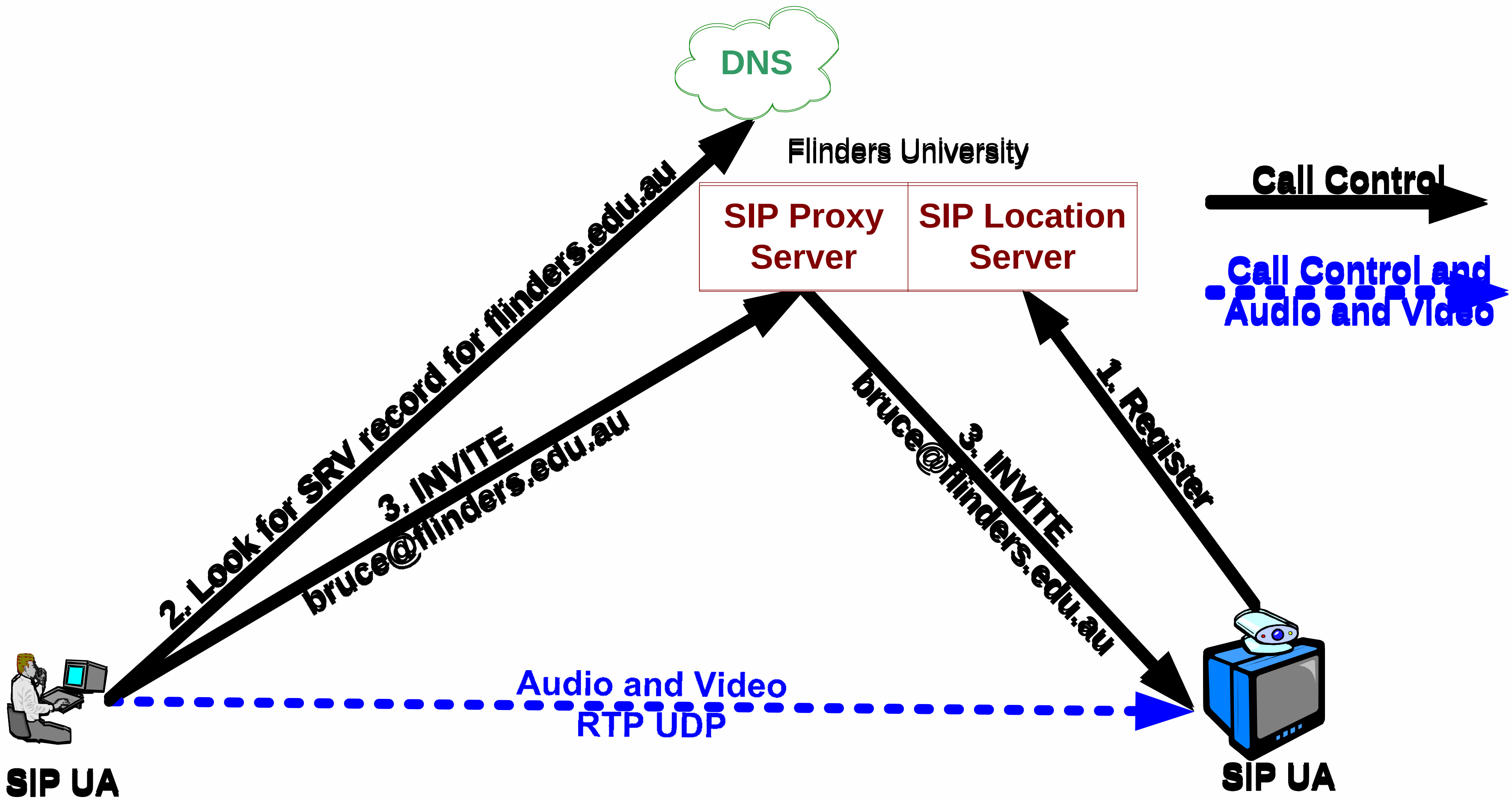
Source RFC 3261

SIP building blocks

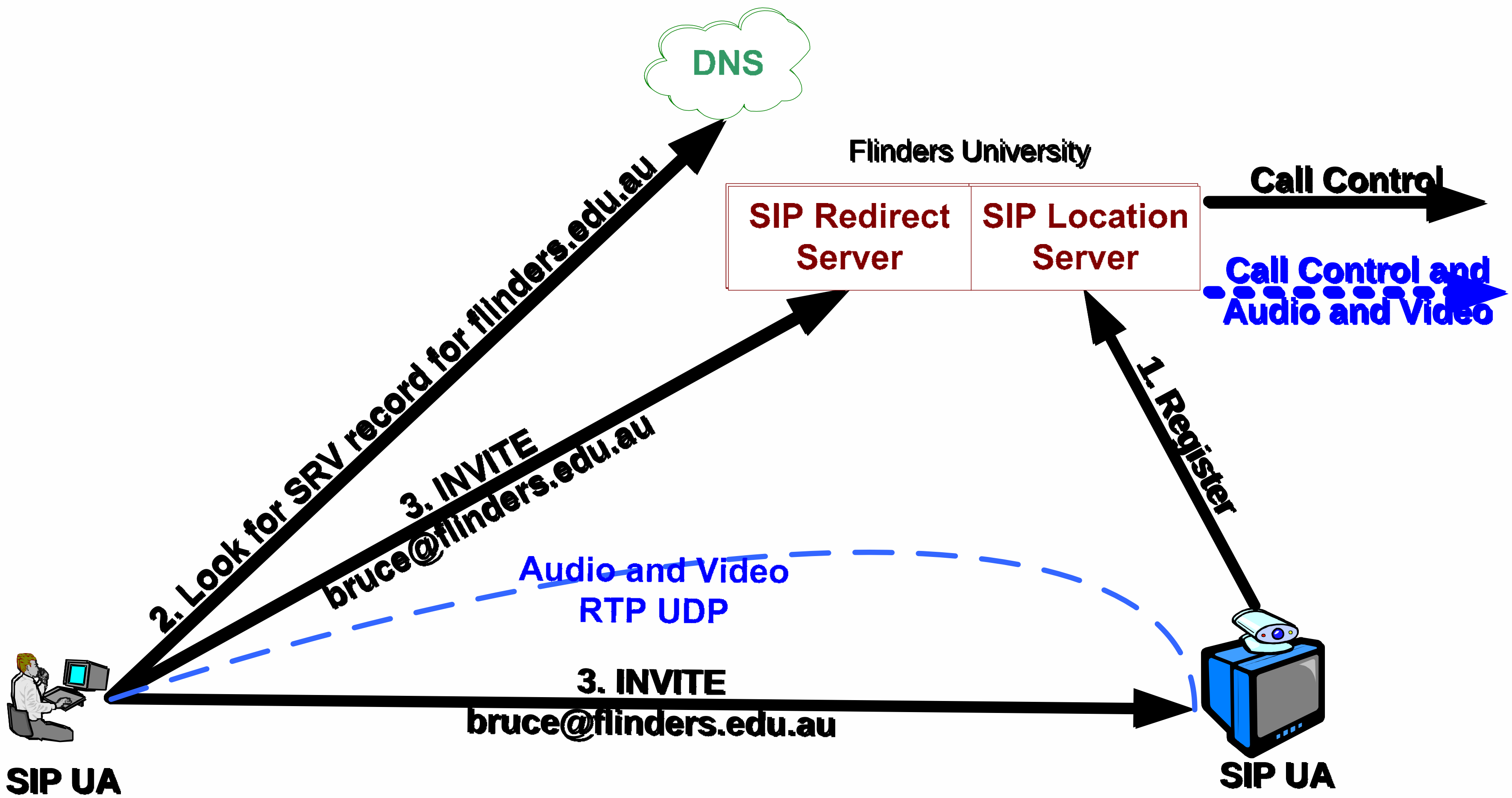
Components:

- User Agents (UAs)
- SIP Location Server
- SIP Redirect Server
- SIP Proxy Server
- SIP Back to Back User Agent (b2bua)
- SIP Gateway

SIP PROXY Server call flow



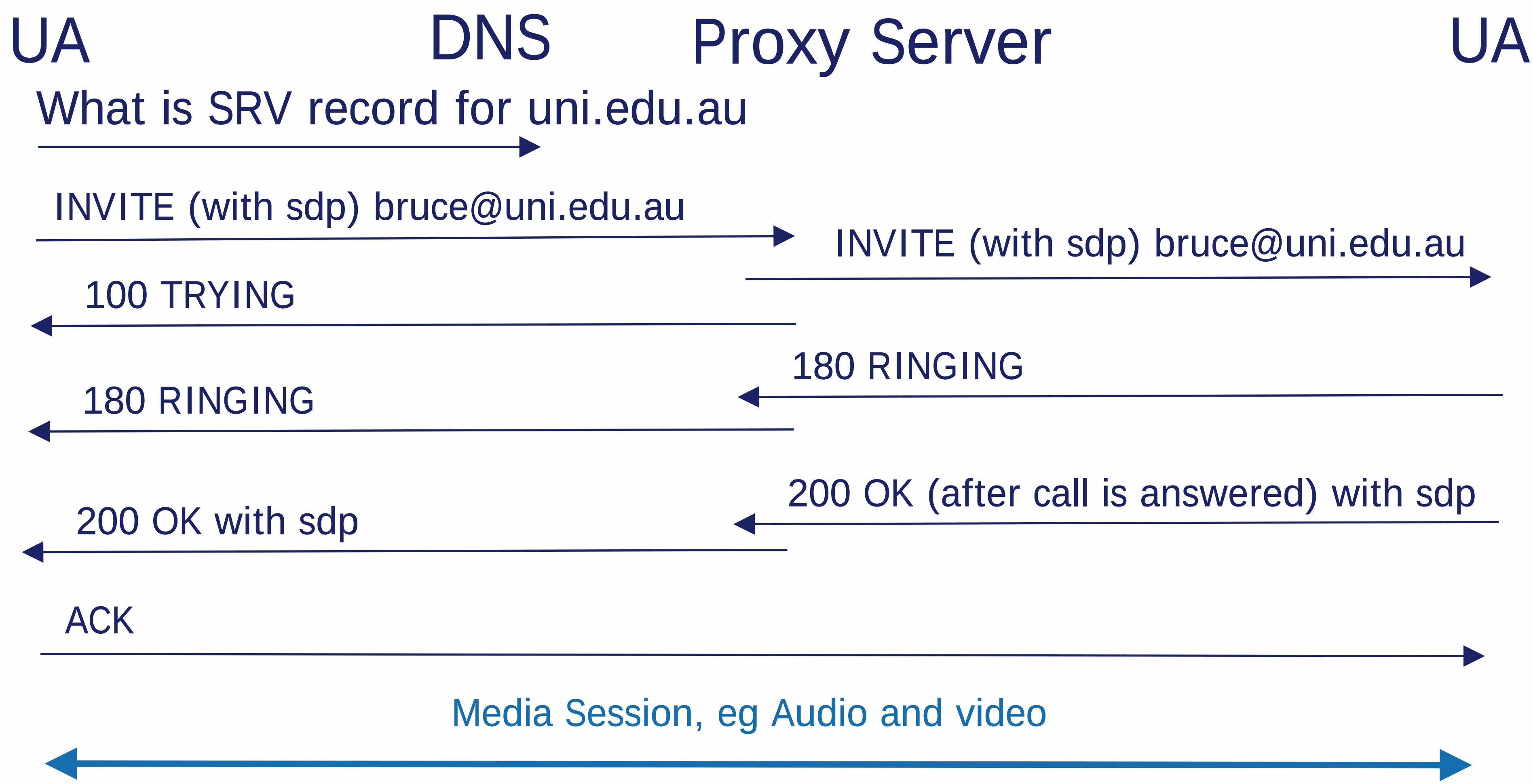
SIP REDIRECT Server call flow



SIP PROXY Server call flow: from RFC3261

```
                                            +----+
                                            | UA |
                                            |    |
                                            +----+
                                               |
                                               |3)INVITE
                                               |   carol@chicago.com
      chicago.com        +--------+            V
      +---------+ 2)Store|Location|4)Query +-----+
      |Registrar|=======>| Service|<=======|Proxy|sip.chicago.com
      +---------+        +--------+=======>+-----+
            A                      5)Resp      |
            |                                  |
            |                                  |
  1)REGISTER|                                  |
            |                                  |
         +----+                                |
         | UA |<-------------------------------+
cube2214a|    |                            6)INVITE
         +----+                    carol@cube2214a.chicago.com
          carol
```

SIP Call flow in more detail (Proxy mode)



Another view from RFC3261

```
                     atlanta.com  . . . biloxi.com
                 .      proxy              proxy     .
               .                                       .
       Alice's  . . . . . . . . . . . . . . . . . . . .  Bob's
      softphone                                        SIP Phone
         |                |                |                |
         |    INVITE F1   |                |                |
         |--------------->|    INVITE F2   |                |
         |  100 Trying F3 |--------------->|    INVITE F4   |
         |<---------------|  100 Trying F5 |--------------->|
         |                |<-------------- | 180 Ringing F6 |
         |                | 180 Ringing F7 |<---------------|
         | 180 Ringing F8 |<---------------|     200 OK F9  |
         |<---------------|    200 OK F10  |<---------------|
         |    200 OK F11  |<---------------|                |
         |<---------------|                |                |
         |                       ACK F12                    |
         |------------------------------------------------->|
         |                   Media Session                  |
         |<================================================>|
         |                       BYE F13                    |
         |<-------------------------------------------------|
         |                     200 OK F14                   |
         |------------------------------------------------->|
         |                                                  |
```

Source RFC 3261

SIP Methods

- REGISTER - login
- INVITE – start a call
- ACK
- CANCEL – abort a call setup
- BYE – end a call
- OPTIONS
- INFO
- REFER - Call Transfer
- MESSAGE - instant messenging
- SUBSCRIBE / NOTIFY - presence

Source RFC 3261

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

How are the codecs negotiated? Answer SDP

- Codecs: the term used to describe the encoding used for the audio and video into data traffic. Eg:
  - Audio G.711 µ-law and A-law, G.723, G.729, etc
  - Video H.261, H.263, H.264, etc
    You could have multiple camera's!
- The sending UA must be able to encode audio/video so that the other end can decode it, and visa versa. "sdp" is the protocol used by the UAs to tell each other what codecs they support.

- SDP is embedded into the SIP Messages.

Presenters note: Review a live call when looking at the following material.

SDP Messages (for reference)

v=0
o=Tesla 289084 289041 IN IP4 lab.high-voltage.org
s=-
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

- v = Protocol Version number (ignored by SIP)
- o = <username> <session id> <version> <network type> <address type>
- <address> (only 3rd field (version) used by SIP)
- s = Session Name <start time> <stop time> (ignored by SIP)
- c = <network type> <address type> <connection address> Connection information (IN =internet, IP4 = IPv4, IP Address)
- t = Time (ignored by SIP)
- m = Media Name and Transport Address (type, port, RTP/AVP Profile) <media> <port> <transport> <fmt list>
- a = Attribute (profile, codec, sampling rate)

SDP Messages (the parts that are interesting)

- **m** <media> <port> <transport> <fmt list>
  Media Name and Transport Address.
  Define all the “audio” capabilities, following by a attributes which have details of each
- a = Attribute (profile, codec, sampling rate). Here is and eg for audio:

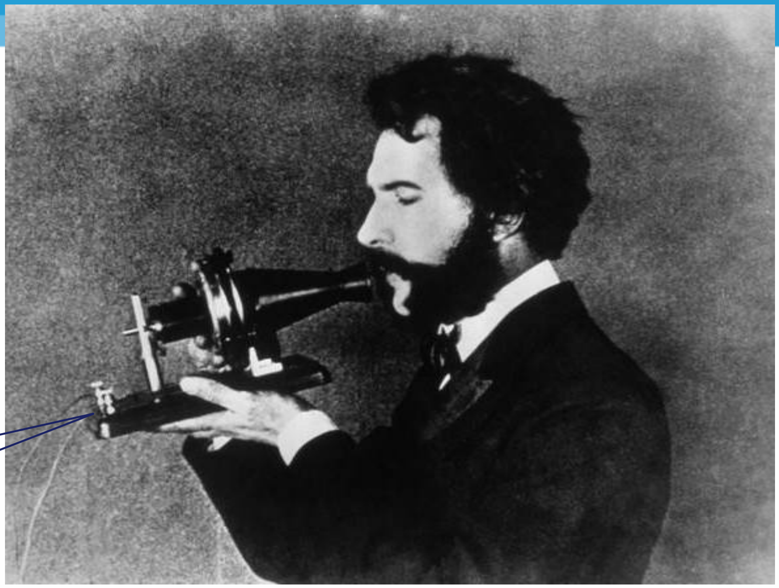
```
m=audio 49170 RTP/AVP 0 8 112
a=rtpmap:0   PCMU/8000
a=rtpmap:8   PCMA/8000
a=rtpmap:112 G7221/16000
```

There could be a second and third m with attributes for video, data, and/or application. Here is one defining Video supported codecs on this UA.

```
m=video 49916 RTP/AVP 34 31
a=rtpmap:34 H263/90000
a=rtpmap:31 H263/90000
```

Source RFC 2327

SIP Addressing in the future will be the preferred address, in addition to Telephone numbers



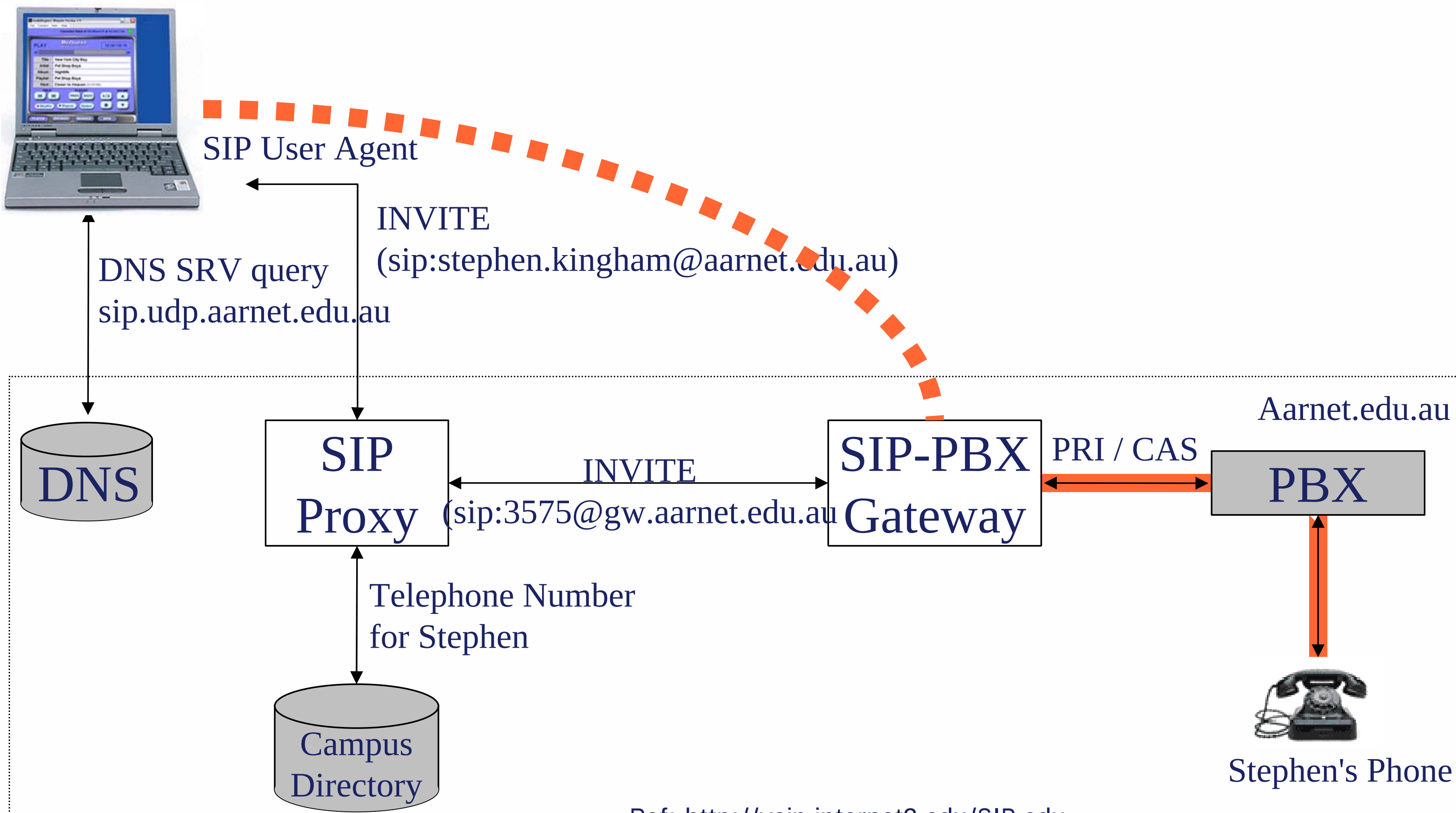
A. G. Bell did **not** say:

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

I will prefer to call people using
sip:Stephen.Kingham@aarnet.edu.au

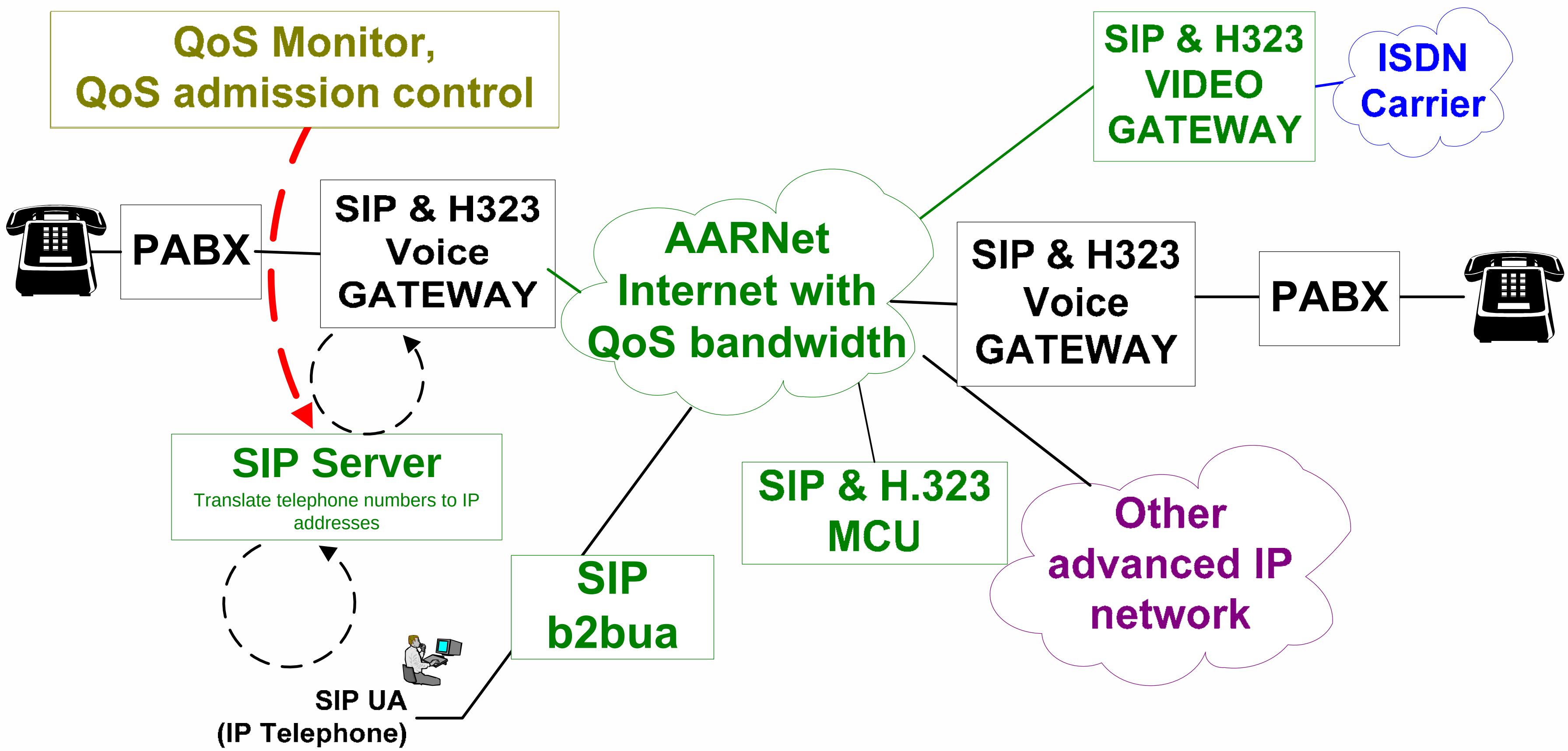
Within the next year you will see this on the bottom of email footers and on business cards of Australian Universities.

SIP.edu Architecture: An achievable goal for a University



Ref: http://voip.internet2.edu/SIP.edu

AARNet SIP & H.323 network
(an example of the building blocks)



SIP History

H.323	SIP
ITU-T protocol	IETF protocol
May 1995	Became “proposed standard” in March 1999.
Study Group 16	Working Groups: SIP, SIPPING, and SIMPLE
Now V.5	Now RFC 3261

from Quincy Wu’s talk, http://www.apan.net Cairns 2004

H323-SIP Comparison of Components

	H.323	SIP
End Station	Terminal	SIP UA
Network Server	Gatekeeper	Registrar, Redirect Server, Proxy Server
	MCU	Conference Server
	PSTN Gateway	PSTN Gateway

from Quincy Wu's talk, http://www.apan.net Cairns 2004

H323-SIP Comparison of Protocols

	H.323	SIP
Signaling	RAS/Q.931	SIP
Capacity Negotiation	H.245	SDP
Codecs	Any	Any
Real-time Communication	RTP/RTCP	RTP/RTCP

from Quincy Wu's talk, http://www.apan.net Cairns 2004

H323-SIP Comparison of Protocols (cont.)

	H.323	SIP
Message Encoding	Binary	ASCII
Transport	UDP and TCP Mostly TCP	UDP and TCP Most UDP
Data Conference	T.120	
Instant Message		RFC 3428
Inter-Domain Routing	Annex G	DNS

from Quincy Wu's talk, http://www.apan.net Cairns 2004

SIP and DNS

- DNS is integral to SIP routing.
- DNS is used to find a priority list of SIP servers for a domain using in SIP specific SRV records into the DNS.
  - Just like MX records in DNS for mail.

- So it turns out it is easy to have backup servers in SIP.

- Good description found on the MIT Internet2 sip.edu project cookbook: http://mit.edu/sip/sip.edu/dns.shtml

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:

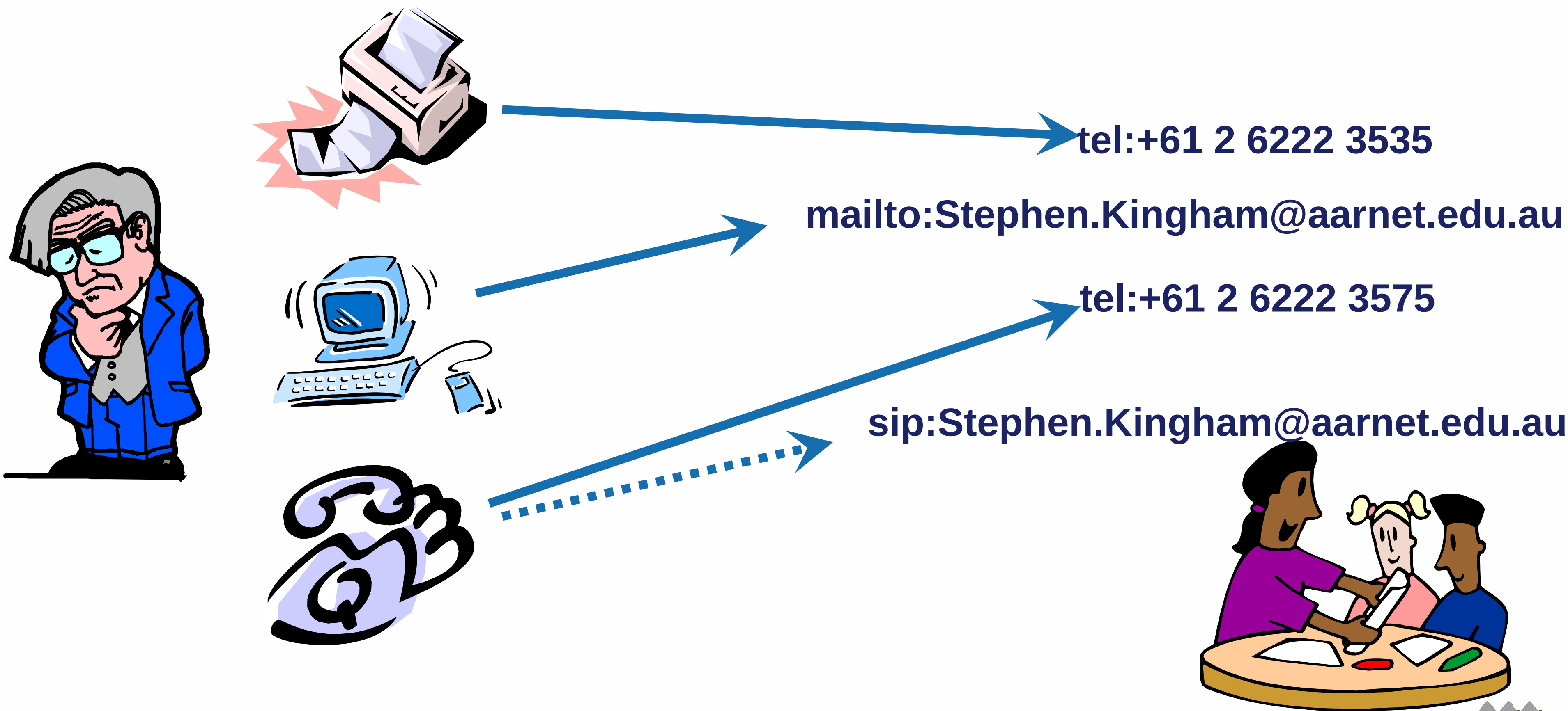
```
;; QUESTION SECTION:
;_sip._udp.aarnet.edu.au.    IN  SRV


;; ANSWER SECTION:
_sip._udp.aarnet.edu.au. 333 IN  SRV 1 1 5060 ser.yarralumla.aarnet.edu.au.
```

SIP and ENUM

- ENUM (rfc 2916) uses the DNS to find the full SIP address using a telephone number. ACA will have ENUM Tier 1 into Australia on 6 June 2005
  http://www.aca.gov.au/telcomm/telephone_numbering/enum_nsg2/.

2. Today, many addresses



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

2. With ENUM, only one

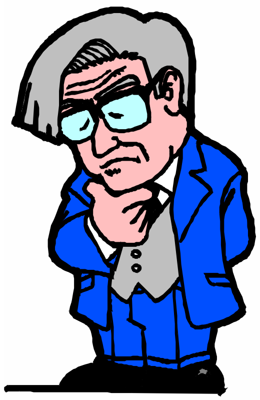
ENUM returns all of these for the caller to choose from:

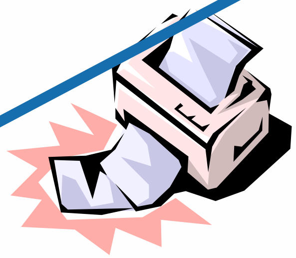
tel:+61 2 6222 3535

mailto:Stephen.Kingham@aarnet.edu.au

tel:+61 2 6222 3575

sip:Stephen.Kingham@aarnet.edu.au



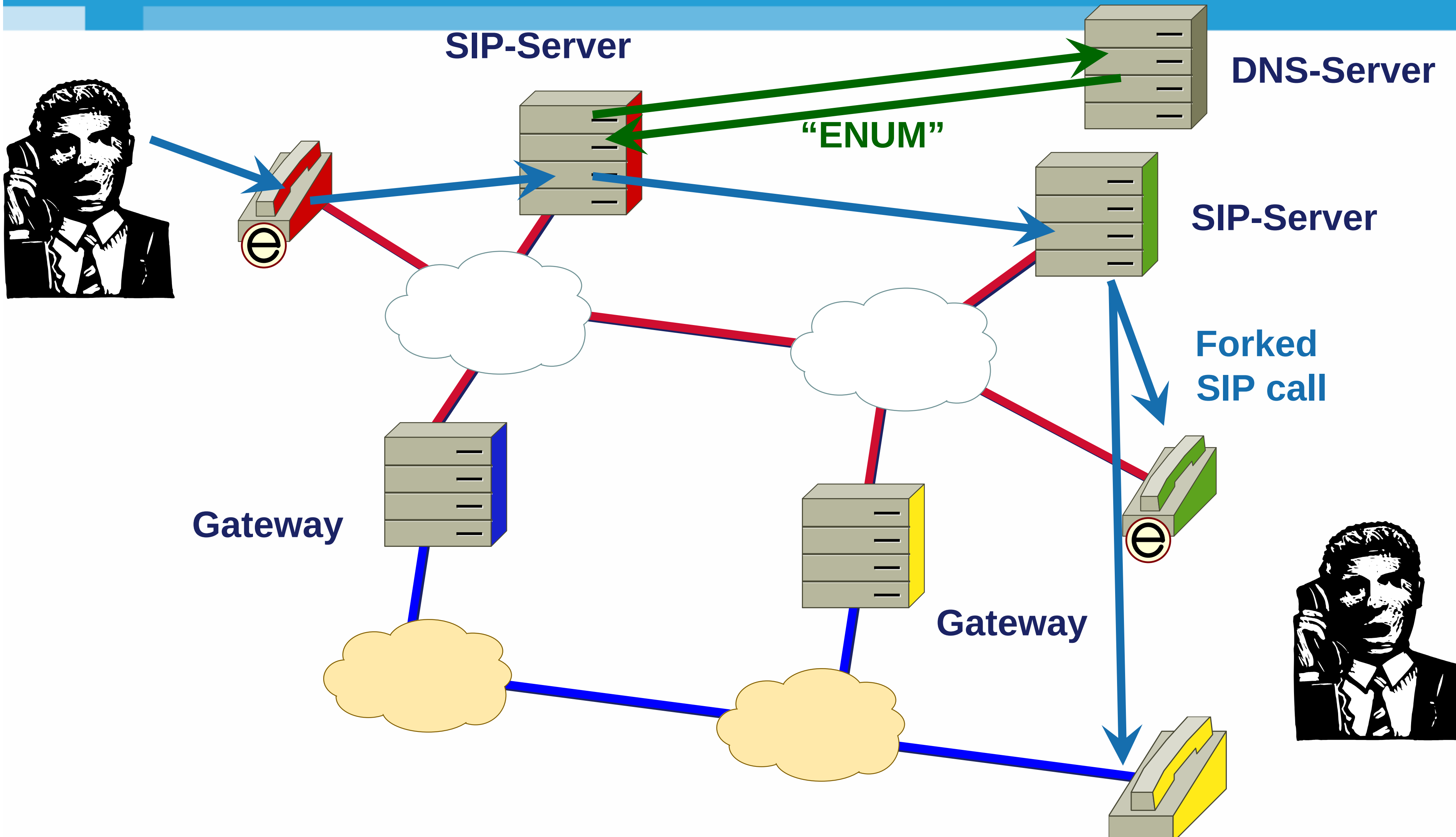


Hand out enum enabled number
+61 2 6222 3575

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

IETF

VoIP via PSTN to PSTN



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

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

IETF

SIP and TRIP (Telephone Routing over IP)

- TRIP (rfc 3219 not passed) does for telephone numbers that BGP does for the entire Internet. Dynamic routing by advertisement!

- More research and experimentation needed here. – perhaps a simpler form of TRIP (STRIP?) encapsulated in MIME?
[Source: Discussions between Randy Bush, Andrew Rutherford and Stephen Kingham 3 Feb 2004].

References used in this talk

- Henning Schulzrinne's Marvelous SIP Page http://www.cs.columbia.edu/sip/
- 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)
- http://www.toyz.org/cgi-bin/sipwiki.cgi

Some more references

- Look up the IETF RFCs on www.ietf.org, or google.
- Good primer: http://www.iptel.org/ser/doc/sip_intro/sip_introduction.html
- Wiki found on www.iptel.org.
- PINT is described in RFC 2458 and RFC 2848

SIP Fault Finding

SIP Workshop
AARNet

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Outline and Objectives

- Review different tools and particular call scenarios
  - Ngrep for unix
  - Ethereal for unix and MicroSoft
  - SIP SAK (SIP Swiss Army Knife)
  - SIPStone
- Server admin:
  - Phpmyadmin to administer mysql data
  - SER specific logs etc
  - Monit and Big Brother
  - DNS
- Review some faults
  - User can not authenticate
  - One way audio

NGREP for Unix

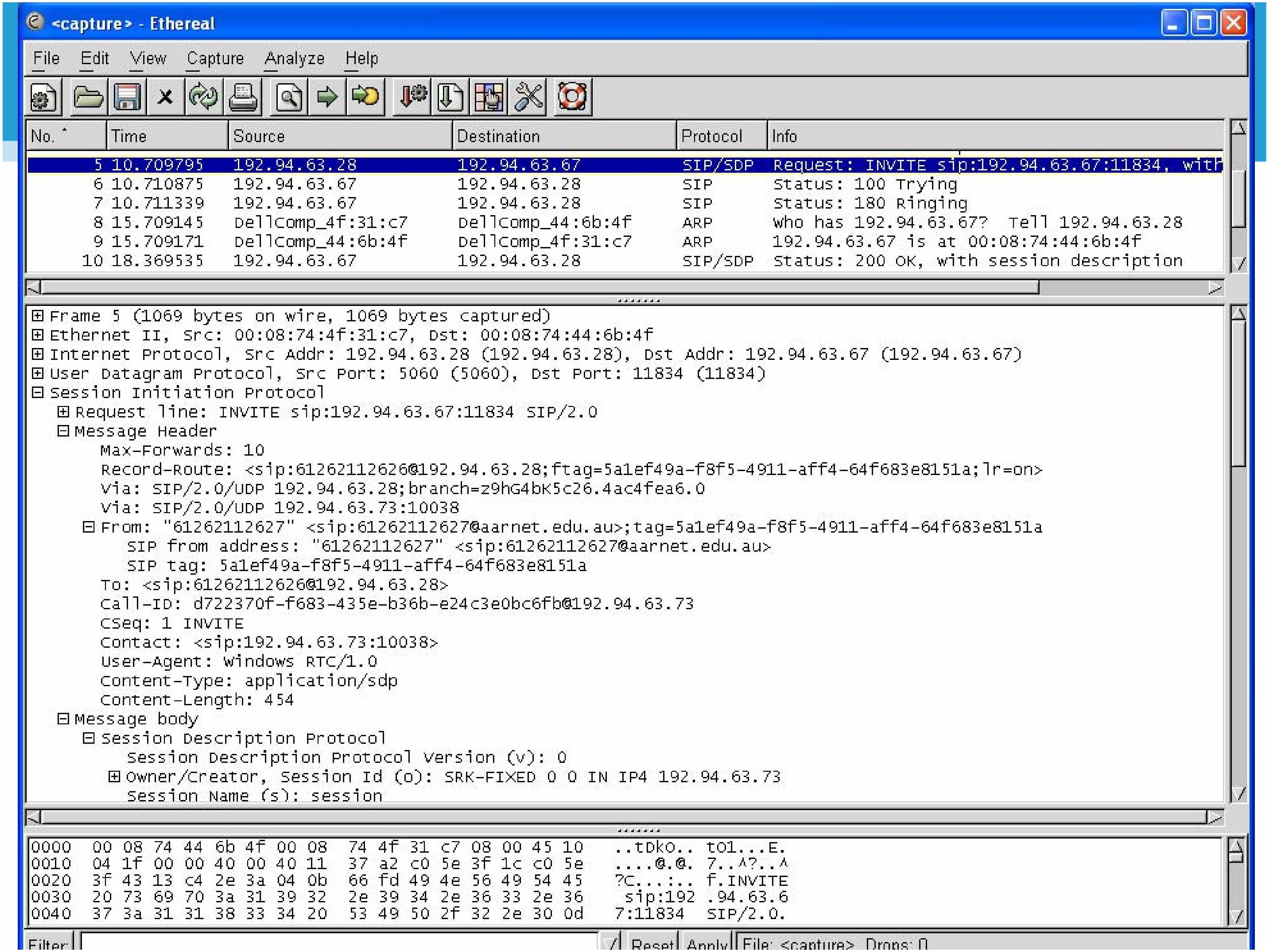
- Perfect for running on SIP Server running on Unix platform.
- Allows you to watch traffic entering leaving the Server.
- Works because SIP protocol is in text and is user readable.
- Get from http://ngrep.sourceforge.net/
- Hint: put in a carriage return at the end of each ".." to make it easier to read.
- **ALWAYS have it running, and keep it for a period of time, eg the last 3 days worth.** That way faults reported after the fact can be investigated. Who in AARNet would like to write a script to do this? This script could get used around the world

Example from NGREP

```
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
```

Ethereal

- Full protocol analyser for Unix, Mac OS, MicroSoft….
- Get ethereal from http://www.ethereal.com/
- For MicroSoft Wondows install WinPcap first from the same location.


<capture> - Ethereal
File Edit View Capture Analyze Help
No. Time Source Destination Protocol Info
5 10.709795 192.94.63.28 192.94.63.67 SIP/SDP Request: INVITE sip:192.94.63.67:11834, with
6 10.710875 192.94.63.67 192.94.63.28 SIP Status: 100 Trying
7 10.711339 192.94.63.67 192.94.63.28 SIP Status: 180 Ringing
8 15.709145 DellComp_4f:31:c7 DellComp_44:6b:4f ARP Who has 192.94.63.67? Tell 192.94.63.28
9 15.709171 DellComp_44:6b:4f DellComp_4f:31:c7 ARP 192.94.63.67 is at 00:08:74:44:6b:4f
10 18.369535 192.94.63.67 192.94.63.28 SIP/SDP Status: 200 OK, with session description
Frame 5 (1069 bytes on wire, 1069 bytes captured)
Ethernet II, Src: 00:08:74:4f:31:c7, Dst: 00:08:74:44:6b:4f
Internet Protocol, Src Addr: 192.94.63.28 (192.94.63.28), Dst Addr: 192.94.63.67 (192.94.63.67)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 11834 (11834)
Session Initiation Protocol
Request line: INVITE sip:192.94.63.67:11834 SIP/2.0
Message Header
Max-Forwards: 10
Record-Route: <sip:61262112626@192.94.63.28;ftag=5a1ef49a-f8f5-4911-aff4-64f683e8151a;lr=on>
Via: SIP/2.0/UDP 192.94.63.28;branch=z9hG4bK5c26.4ac4fea6.0
Via: SIP/2.0/UDP 192.94.63.73:10038
From: "61262112627" <sip:61262112627@aarnet.edu.au>;tag=5a1ef49a-f8f5-4911-aff4-64f683e8151a
SIP from address: "61262112627" <sip:61262112627@aarnet.edu.au>
SIP tag: 5a1ef49a-f8f5-4911-aff4-64f683e8151a
To: <sip:61262112626@192.94.63.28>
Call-ID: d722370f-f683-435e-b36b-e24c3e0bc6fb@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: 454
Message body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): SRK-FIXED 0 0 IN IP4 192.94.63.73
Session Name (s): session
0000 00 08 74 44 6b 4f 00 08 74 4f 31 c7 08 00 45 10 ..tDkO.. tO1...E.
0010 04 1f 00 00 40 00 40 11 37 a2 c0 5e 3f 1c c0 5e ....@.@. 7..^?..^
0020 3f 43 13 c4 2e 3a 04 0b 66 fd 49 4e 56 49 54 45 ?C...:.. f.INVITE
0030 20 73 69 70 3a 31 39 32 2e 39 34 2e 36 33 2e 36 sip:192 .94.63.6
0040 37 3a 31 31 38 33 34 20 53 49 50 2f 32 2e 30 0d 7:11834 SIP/2.0.
Filter: Reset Apply File: <capture> Drops: 0

A Grab bag of tools

- SIP SAK – SIP Swiss Army Knife
  - A "traceroute" like tool for SIP.
  - Get from http://sipsak.berlios.de/
- SIPStone
  - Measuring SIP performance from Columbia University USA.
  - See http://www.sipstone.com/ and/or http://www.cs.columbia.edu/IRT/cinema/sipstone
- MONIT and Big Brother
  - Monit is a server watching utility which alerts administrators when a server dies.
  - Big Brother is another server watching utility which alerts administrators when a server dies.
- Phpmyadmin – to manage mysql tables
  - Excellent for managing the SER Proxy Server data stored in the mysql data base.
  - Get it from www.phpmyadmin.net.

DNS

- SIP relies on DNS for routing (eg finding other SIP Servers). If something goes wrong with DNS then call setups can block for several seconds. Mitigate by:
  - Cache DNS (eg nscd daemon in Linux)
  - Have plenty of free children (threads) in the Proxy Server
  - Process transactions statefully to absorb retransmissions without additional DNS lookups.

Hot topis

- Slide 67