SIP Tutorial

→VoIP Workshop
Terena 2005 Poznan Poland

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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
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

<table>
<thead>
<tr>
<th>Feature</th>
<th>RFC Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>call waiting/multiple calls</td>
<td>RFC 3261</td>
</tr>
<tr>
<td>hold</td>
<td>RFC 3264</td>
</tr>
<tr>
<td>transfer</td>
<td>RFC 3515/Replaces</td>
</tr>
<tr>
<td>conference</td>
<td>RFC 3261/callee caps</td>
</tr>
<tr>
<td>message waiting</td>
<td>message summary package</td>
</tr>
<tr>
<td>call forward</td>
<td>RFC 3261</td>
</tr>
<tr>
<td>call park</td>
<td>RFC 3515/Replaces</td>
</tr>
<tr>
<td>call pickup</td>
<td>Replaces</td>
</tr>
<tr>
<td>do not disturb</td>
<td>RFC 3261</td>
</tr>
<tr>
<td>call blast</td>
<td>RFC 3261</td>
</tr>
</tbody>
</table>

### Boss/Admin features

<table>
<thead>
<tr>
<th>Feature</th>
<th>RFC Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>simultaneous ringing (forking)</td>
<td>RFC 3261</td>
</tr>
<tr>
<td>basic shared lines</td>
<td>dialog/reg. package</td>
</tr>
<tr>
<td>barge-in</td>
<td>Join</td>
</tr>
<tr>
<td>“Take”</td>
<td>Replaces</td>
</tr>
<tr>
<td>Shared-line “privacy”</td>
<td>dialog package</td>
</tr>
<tr>
<td>divert to admin</td>
<td>RFC 3261</td>
</tr>
<tr>
<td>intercom</td>
<td>URI convention</td>
</tr>
<tr>
<td>auto attendant</td>
<td>RFC 3261/2833</td>
</tr>
<tr>
<td>attendant console</td>
<td>dialog package</td>
</tr>
<tr>
<td>night service</td>
<td>RFC 3261</td>
</tr>
</tbody>
</table>

### Attendant features

<table>
<thead>
<tr>
<th>Feature</th>
<th>RFC Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>privacy</td>
<td></td>
</tr>
<tr>
<td>barge-in</td>
<td></td>
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<tr>
<td>basic shared lines</td>
<td></td>
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<td>divert to admin</td>
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<td>auto attendant</td>
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<tr>
<td>attendant console</td>
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<tr>
<td>night service</td>
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</tbody>
</table>

From Rohan Mahy’s VON Fall 2003 talk
Australia’s Academic and Research Network

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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!!!!
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

1. Register SIP UA
2. Look for SRV record for flinders.edu.au
3. INVITE bruce@flinders.edu.au

SIP Proxy Server | SIP Location Server

DNS

Call Control

Call Control and Audio and Video

Audio and Video
RTP UDP

SIP UA

Flinders University

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SIP REDIRECT Server call flow

1. Register
   Flinders University

2. Look for SRV record for flinders.edu.au

3. INVITE bruce@flinders.edu.au

DNS

1. Register

SIP Redirect Server
SIP Location Server

Call Control

Call Control and Audio and Video

Audio and Video RTP UDP

SIP UA

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SIP PROXY Server call flow: from RFC3261

+----+ 1) REGISTER   A  5) Resp
 | UA |  |
 |    |  |
 |-----|  |
 | 3) INVITE  carol@chicago.com |
|      |  |
|      |  v
chicago.com +-----+ 2) Store | Location | 4) Query  +-----+
| Registrar | ======> | Service |<========| Proxy | sip.chicago.com
| A   +-----+  +-----+  +-----+
|  |
| 1) REGISTER |
|  |
|  |
|  |
| 6) INVITE  carol@cube2214a.chicago.com |
cube2214a |  |
|  |
|  |
|  |
carol
SIP Call flow in more detail (Proxy mode)

What is SRV record for uni.edu.au

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

100 TRYING

180 RINGING

200 OK with sdp

ACK

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

180 RINGING

200 OK (after call is answered) with sdp

ACK

Media Session, eg Audio and video

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Another view from RFC3261

```
atlanta.com . . . biloxi.com
          . proxy              proxy .
          .      proxy              proxy .
Alice's . . . . . . . . . . . . . . . . . . . .  Bob's
softphone                                           SIP Phone

<p>| | | |
|                |                |                |
|    INVITE F1   |                |                |
|&lt;----------------|    INVITE F2   |                |
|  100 Trying F3 |---------------&gt;|    INVITE F4   |
|&lt;----------------|  100 Trying F5 |---------------&gt;|
|                   |  180 Ringing F6 |              |
|                   |  180 Ringing F7 |&lt;---------------|
|  180 Ringing F8  |&lt;---------------|     200 OK F9  |
|&lt;----------------|                |&lt;---------------|
|  200 OK F10     |                |&lt;---------------|
|&lt;----------------|                |                |</p>
<table>
<thead>
<tr>
<th></th>
<th>ACK F12</th>
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<tbody>
<tr>
<td>Media Session</td>
<td></td>
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<tr>
<td>&lt;----------------</td>
<td>----------------------</td>
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<td>&lt;----------------</td>
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<td></td>
</tr>
<tr>
<td></td>
<td>200 OK F14</td>
</tr>
</tbody>
</table>
```
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 messaging
- 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.
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:

  \[
  \begin{align*}
  \text{m} &= \text{audio} \ 49170 \ \text{RTP/AVP} \ 0 \ 8 \ 112 \\
  \text{a} &= \text{rtpmap:0} \ \text{PCMU/8000} \\
  \text{a} &= \text{rtpmap:8} \ \text{PCMA/8000} \\
  \text{a} &= \text{rtpmap:112} \ \text{G7221/16000}
  \end{align*}
  \]

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.

  \[
  \begin{align*}
  \text{m} &= \text{video} \ 49916 \ \text{RTP/AVP} \ 34 \ 31 \\
  \text{a} &= \text{rtpmap:34} \ \text{H263/90000} \\
  \text{a} &= \text{rtpmap:31} \ \text{H263/90000}
  \end{align*}
  \]
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!”
```

© Ben Teitelbaum @ Internet2

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)

- **QoS Monitor, QoS admission control**
- **SIP & H.323 Voice GATEWAY**
- **SIP Server**
  - Translate telephone numbers to IP addresses
- **SIP b2bua**
- **SIP & H.323 MCU**
- **SIP & H.323 VIDEO GATEWAY**
- **ISDN Carrier**
- **PABX**

Internet with QoS bandwidth

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## SIP History

<table>
<thead>
<tr>
<th>H.323</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>ITU-T protocol</td>
<td>IETF protocol</td>
</tr>
<tr>
<td>Study Group 16</td>
<td>Working Groups: SIP, SIPPING, and SIMPLE</td>
</tr>
<tr>
<td>Now V.5</td>
<td>Now RFC 3261</td>
</tr>
</tbody>
</table>

From Quincy Wu’s talk, [http://www.apan.net](http://www.apan.net) Cairns 2004
<table>
<thead>
<tr>
<th></th>
<th>H.323</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>End Station</strong></td>
<td>Terminal</td>
<td>SIP UA</td>
</tr>
<tr>
<td><strong>Network Server</strong></td>
<td>Gatekeeper</td>
<td>Registrar, Redirect Server, Proxy Server</td>
</tr>
<tr>
<td><strong>MCU</strong></td>
<td></td>
<td>Conference Server</td>
</tr>
<tr>
<td><strong>PSTN Gateway</strong></td>
<td></td>
<td>PSTN Gateway</td>
</tr>
</tbody>
</table>

from Quincy Wu’s talk, [http://www.apan.net](http://www.apan.net) Cairns 2004
### H.323-SIP Comparison of Protocols

<table>
<thead>
<tr>
<th></th>
<th>H.323</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Signaling</td>
<td>RAS/Q.931</td>
<td>SIP</td>
</tr>
<tr>
<td>Capacity Negotiation</td>
<td>H.245</td>
<td>SDP</td>
</tr>
<tr>
<td>Codecs</td>
<td>Any</td>
<td>Any</td>
</tr>
<tr>
<td>Real-time Communication</td>
<td>RTP/RTCP</td>
<td>RTP/RTCP</td>
</tr>
</tbody>
</table>

from Quincy Wu’s talk, [http://www.apan.net](http://www.apan.net) Cairns 2004
<table>
<thead>
<tr>
<th></th>
<th>H.323</th>
<th>SIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Message Encoding</td>
<td>Binary</td>
<td>ASCII</td>
</tr>
<tr>
<td>Transport</td>
<td>UDP and TCP Mostly TCP</td>
<td>UDP and TCP Most UDP</td>
</tr>
<tr>
<td>Data Conference</td>
<td>T.120</td>
<td></td>
</tr>
<tr>
<td>Instant Message</td>
<td>RFC 3428</td>
<td></td>
</tr>
<tr>
<td>Inter-Domain Routing</td>
<td>Annex G</td>
<td>DNS</td>
</tr>
</tbody>
</table>

from Quincy Wu’s talk, [http://www.apan.net](http://www.apan.net) Cairns 2004
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](http://mit.edu/sip/sip.edu/dns.shtml)
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.yarralumla.aarnet..edu.au. IN A 192.94.63.28
ser.nsw.aarnet..edu.au. IN A 138.44.16.90
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.
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
2. Today, many addresses

tel:+61 2 6222 3535
mailto:Stephen.Kingham@aarnet.edu.au
tel:+61 2 6222 3575
sip:Stephen.Kingham@aarnet.edu.au

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
VoIP via PSTN to PSTN

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

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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
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
• 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

By Stephen Kingham
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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=484b98c9-2db44512-95f9-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
• Full protocol analyser for Unix, Mac OS, Microsoft…..
• Get ethereal from [http://www.ethereal.com/](http://www.ethereal.com/)
• For Microsoft Windows install WinPcap first from the same location.
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.
• 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.
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 nsqd daemon in Linux)
- Have plenty of free children (threads) in the Proxy Server
- Process transactions statefully to absorb retransmissions without additional DNS lookups.
Hot topics

• Slide 67