# SIP Routing in a Nutshell

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IETF 83, Routing Area Meeting
March 2012

v1

# SIP Identifiers

- Email style names:
  - sip:fluffy@example.com
- Telephone number style names:
  - tel:+1-408-555-1212
  - sip:+14085551212@example.com;user=phone

# Forwarding

- Number based routing
  - +1-408-*  forward to level3.com
- End users forwards phone to another number
  - +1-408-555-1212 forward to +1-408-555-1234
- Nexthop domain lookup based on DNS SRV / NAPTR

- Have a BGP like protocol, TRIP (RFC 3219) with little or no usage
- Many route tables are managed with Excel spreadsheets

# Return Path Routing

- Initial requests keep a via list of nodes traversed
- Responses routed by traversing the VIA list

- Original design had node insert it's own address in VIA
- Moved to having the node that received a message insert the address of where it perceived the messages had come from
- Works better with NATs

# Pre-Loaded Route Header

- Proxies (the routing element) can add a header like
  - Route: <sip:example.com;lr>
- to force routing through example.com

# Dynamic Pre-Loaded Route

- Proxies (the routing element) can add a header like
  - Record-Route: <sip:example.com;lr>
- to force routing through example.com on future messages in the same "phone call"
- Used for wide variety of reasons including nailing down future messages to go through same routing node in a cluster as the one keeping state for this dialog

# Rendezvous

- A phones registers is contact information with registrar this creates a mapping from the users Address of Record such as fluffy@example.com to the current IP address of the phone.
- This is an ID / Locator split. The ID is the SIP Address Of Record (AOR) and locator is path to phone from the registrar.
- If a phone needs incoming messages to be routed via a particular proxy, called p1, the locator might look like:

  Contact: <sip:line1@192.0.2.4:5077>, Path: <sip:example.com;lr>

- Can work with complex NAT / Firewall topologies.

# Pre-Loaded Path Header

- Can add a header like
    - Path: <sip:example.com;lr>
- to tell the rendezvous point to force future calls thought example.com
- Used for discovery of devices that future calls needs to be forced through

# Forwarding Metrics

- Often least cost routing where cost is $$$
  - add in additional constraints to ensure adequate call completion and voice quality metrics and maintained
  - complications on what happens when come to the end of block of minutes
  - cross boarder tariffs result in requirements for minute smuggling

# Loop Detection

- Two approaches (belts & suspenders)
  - Time To Live counter
  - VIA hop labels - each hop adds a via value and can look at previous via list to detect a loop
- Spirals
  - Consider a call from a@a.com to B@b.com who forward to c@a.com. This traverses a.com twice but that is not an error.

# Topology hiding

- SP A in say the US hands all it’s call outside US to SP B. For calls to kenya, lets says SP B hands the call to SP C in kenya.
- Often SP B does not want to reveal to SP A that it used SP C to terminate the call.
- Solution: a SIP layer NAT aka SBC (Session Boarder Controller) or B2BUA. (Back to Back User Agent)
  - This will often trash the TTL counter
  - Usually remove the VIA stack
- Can end up breaking loop detection