Addressing Record-Route issues in Session Initiation Protocol (SIP)

(draft-froment-sip-record-route-fix-00.txt)

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Why this draft?

- Based on implementor experience in SIP interoperability events (SIPIT) in the last three years

1/ Deprecate record-route rewriting, and formally suggest to recommend double record-routing.

2/ Clarify RFC 3261 scenarios on Record-Route: bad implementation choices, IP address versus logical names in RR, transport switching, multi-homed use cases...

What is the problem? 1/3

1/ **Rewriting is bad**

Route seen by the caller is different from the Route seen by the callee

- Callee cannot sign the route set, because it gets edited by the proxy in the response. Consequently, end-to-end protection of the route set can not be supported by the protocol. The openness and the end-to-end principles are broken..
- Proxy must implement special "multi-homed" stateful logic. On the request phase, it goes through output interface calculation and writes the output interface into the route.

What is the problem? 2/3

2/ **Double record-routing is good**, BUT, its specification is spread in multiple documents, none of them handling the general use case in core spec.

- [**RFC3486**], describes the double Record-Routing as an alternative to the record-route rewriting in responses. This document is limited in scope to the "comp=sigcomp" parameter when doing compression with SIGCOMP.
- [**RFC3608**], recommends the usage of double Record- Routing instead of the rewriting solution described in [RFC3261] for "Dual-homed" proxies.
- ID [**draft-ietf-sipping-v6-transition-04**], mandates double Record-Routing for multi-homed proxies doing IPV4/ IPV6 transitions, when proxy inserts IP addresses.
- ID [**draft-ietf-sip-sips-01**], recommends to apply the double Record-Routing technique when a proxy has to change the scheme from sip to sips; again, the scope is limited to this use case.

Consequence: **some** implementors don’t even know it exists!

What is the problem? 3/3

3/ Very basic interworking between UAs and SIP proxies are still very often not working at SIPIT, e.g.:

- Alice UA calls Bob UA though company LAMBDA proxy.
- Alice call bob in TCP, proxy switches to UDP since Bob is registered in UDP.
- Proxy puts a Record-Route with NO transport parameter (RFC 3261, 16.6 The URI SHOULD NOT contain the transport parameter unless the proxy has knowledge (such as in a private network) that the next downstream element that will be in the path of subsequent requests supports that transport.)

◊ Alice switches from TCP to UDP when sending its ACK (no transport param ∫ UDP): this is an unwanted behavior...

 ◊ Solution: IP Address should not be used in Record-Route, a logical name should be put in RR, and UAs should use NAPTR/DNS (3263) to find the right transport.

 ◊ Some implementation still want to use IP, and/or some UAs don't do NAPTR (still around 50/60% of implementations)... The transport switching can still occur when UDP datagram exceeds MTU size..

 So, some proxies choose to always put transport parameter AND double record-route: this MAY be problematic if downstream element that will be in the path of subsequent requests does not support a non-mandatory transport (SCTP?).

4/ Other problematic scenarios: general multi-homed proxy use case, sip/sips (ok, this one will be fixed in sip-sips draft...)

Next? 1/2

- Proposed standard or BCP?
  - Rewording some sections of 3261 to deprecate rewriting and/or suggest double-record- routing as an alternative is clearly a normative change.
  - Clarifying the multi-homed and transport switching scenarios is closer to a BCP, even if some rewording of 3261 could be useful.

Next? 2/2

- Positive feedback from reviewers:
  - Few open issues:
    - should better distinguish bcp aspects from normative aspects,
    - Improve bcp to cover all use cases,
    - security section to be improved,...
    - but not a lot of work remaining...
- Can be fixed quickly without waiting for RFC 3261bis or « SIP 3.0 » ;-) ...
- WG item?