SIP Interoperability Experiences and Recommendations

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Input collected inside Nokia from people involved in SIP testing and deployment cases

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SIP interoperability today as we see it

- Pick two SIP UAs (at random, using <u>default configuration</u>)
- Connect them by picking a couple of proxies, registrars, SBCs, ALGs (again at random)
- Only use products that claim to support "SIP 2.0"

  - **<u>Probability that ALL features interoperate between the UAs is <50%</u>**

- And this is only about voice calls and a few additional telephony features
  - SIMPLE will cause additional headache (although encouraging results at SIPit 21!)
  - ICE will be absolutely needed but another huge interop challenge
  - BLISS, XCON, … still too far to determine
- Service providers need to choose their equipment carefully, and write their own compliancy requirements
  - These are often contrary to IETF standards to make things work
  - Huge amount of extra work for UA vendors
    - (Auto-) Configurability of UAs important to configure which hacks to support…

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Roots of the Problems?

- Too many separate RFCs containing essential features/corrections
  - High risk that not everyone supports all “important” ones
- Too much flexibility. RFCs as such are correct, but allow too many possibilities – takes too much effort to implement and <u>test</u> everything
  - SDP offer/answer (by far the most problematic!)
  - Call transfer
  - Message routing within a dialog
- Some key features are still missing
  - User Agent configuration and outbound proxy discovery
- Competing standards
  - IETF vs. 3GPP: Registration logic, Security
  - Different profiles (without proper feature negotiation)

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What can be done?

- Feature set recommendations
  - Define meaningful sets/modules out of the dozens of RFCs
  - SIP Forum started this but needs more cycles and commitment
- Best practices documents
  - Describe how to make things work in a robust manner
  - Define conventions for things that are left open in the standards
- Limiting flexibility
  - Is this anymore possible for e.g. SDP offer/answer?
- Strong links between SIPit and IETF
  - Pay attention to poor interop results from SIPit
  - Good discussion started based on SIPit 21!
- Proof-of-concept open source implementations
- Certification program
  - Guarantee that at least some basic feature set works
  - Help service providers to get compatible products

Further recommendations

- Drop RFC 2543 backwards compatibility
- Use TCP-only
  - No transport selection
  - NAT traversal becomes easier
  - No issues with message size and IP-level fragmentation
  - Why do we insist requiring UDP?
- Focus on the most critical issues
  - We need <u>working</u> UA configuration and SIP-outbound solutions
    - What really has to work? Is there anything that can be taken away?
  - Fix SDP negotiation and NAT traversal
- SIP 3.0
  - Include all essential corrections (NAT traversal, etc.)
  - Leave out bad ideas (UDP, etc.)

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Typical interoperability issues we have seen (1/2)

- SDP offer/answer
  - Still the number one problem area in SIPits and also in the field
  - Too much flexibility – implementer's are not able to support all possible cases
    - SDP offer and answer can be carried in various ways in SIP requests/responses
    - Offerless INVITE, 183/PRACK, ...
    - Codec order, dynamic payload numbers
  - Dangerous to support exotic features as the risk of non-interoperability increases
    - Example: We implemented the trick described in RFC 3264 Section 10.2 related to nailing down a single codec. This caused so many interop issues that we were forced to abandon it
  - Even if UAs understand each other, B2BUAs in the middle get confused...
  - Any possibility to limit the flexibility should be pursued
  - Publish some kind of SDP offer/answer with SIP cookbook
- Preconditions
  - Many (IMS-oriented) UAs do not implement the actual mechanism, just the single call-flow
  - Need tricks to recover
- Digest authentication
  - Relation of digest Username authentication Realm to SIP AoR causes confusion
  - Recommend how to derive these from the AoR as the default?

Typical interoperability issues we have seen (2/2)

- user=phone parameter
  - Inclusion or omission will lead to '404' depending on the environment
    - We needed to make this yet another configurable parameter
  - Some clarification still needed
- Message routing
  - Still very common that proxies/B2BUAs do not support loose routing, they either reject requests carrying Route header or instead of record-routing themselves they overwrite Contact URI
  - UAs do various hacks to recover from errors, e.g. ignore lost ACK
- NAT traversal
  - Transport selection for UA behind NAT does not work, has to be forced to be either TCP or UDP
  - Proxies are not able to use UA-initiated TCP connections for incoming requests
  - SIP-Outbound should fix these problems in the long run, but includes a lot of additional features that would not be necessary
  - Use of TCP-only would solve many problems

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