SIP Operation in the Public Internet

An Update on What Makes Running SIP a Challenge and What it Takes To Deal With It

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Outline

• Status update: where iptel.org’s operational experience comes from and what works today
• Trouble-stack: things which do not fly yet
• Operational Practices
• Conclusions
Background

• iptel.org has been running SIP services on the public Internet since 2001. Users are able to pick an address username@iptel.org and a numerical alias.
• The infrastructure serves public subscribers as well as internal users with additional privileges (PSTN termination, voicemail).
• Services powered by open-source SIP server, SIP Express Router (ser).
• Increase in population size since introduction of Windows Messenger: free Microsoft SIP client with support for VoIP, video, instant messaging and collaborative applications.
Good News …

• Basic VoIP services work, so do complementary integrated services such as instant messaging, voicemail, etc.
  – Commercial deployments exist, mostly offering PSTN termination: Vonage, deltathree, denwa, Packet 8
  – Trial services: FWD, PCH, WCOM, SIP Center
  – Tens of intranet deployment of SER reported, probably many more unknown

• Billing machinery works too: Accounting easy, though not standardized.

• Numbering plans easy to maintain and they complement domain names well.
… Good News

- QoS mostly pleasant for broadband community:
  - *Links between iptel.org site and iptel.org user community have packet loss close to zero and RTT mostly bellow 150 ms, rarely above 200 ms.*

- SIP interoperability well established across mature implementations

- Interoperation with other technologies works too:
  - *Competition on the PSTN gateway market established*
  - *Gateway to Jabber instant messaging up and running*
  - *Commercial H.323 gateways exist*
Bad News

- Nightmare – NATs (…)
- Why I keep my PSTN black phone in my room’s corner: Reliability (…)
- What Is It? Machines Do, Operators Don’t … Scalability (…)
- End-devices still expensive
- Future issues: spam, denial of service attacks
NAT Traversal

• NATs popular because they conserve IP address space and help residential users to save money charged for IP addresses.

• Problem: SIP does not work over NATs without extra effort. Peer-to-peer applications’ signaling gets broken by NATs: Receiver addresses announced in signaling are invalid out of NATted networks.

• Straight-forward solution: IPv6 – unclear when deployed if ever.

• There are many scenarios for which no single solution exists (they primarily differ in design properties of NATs – symmetric, app-aware, etc.)
Current NAT Traversal Practices …

• Application Layer Gateways (ALGs) – built-in application awareness in NATs.
  – Requires ownership of specialized software/hardware and takes app-expertise from router vendors (Intertex, PIX).

• Geeks’ choice: Manual configuration of NAT translations
  – Requires ability of NATs, phones, and humans to configure static NAT translation. (Some have it.) If a phone has no SIP/NAT configuration support, an address-translator can be used.

• UPnP: Automated NAT control
  – Requires ownership of UPnP-enabled NATs and phones. NATs available today, phones rarely (Snom).
… Current NAT Traversal Practices

• STUN: Alignment of phones to NATs
  – Requires NAT-probing ability (STUN support) in end-devices and a simple STUN server. Implementations exist (snom, kphone).
  – Does not work over NATs implemented as “symmetric”.
  – Troubles if other party in other routing realm than STUN server.
+ Works even if NAT device not under user’s control.

• Relay: Each party maintains client-server communication
  – Introduces a single point of failure; media relay subject to serious scalability and reliability issues
+ Works over most NATs
## NAT Practices: Overview

<table>
<thead>
<tr>
<th></th>
<th>ALG</th>
<th>STUN</th>
<th>UPnP</th>
<th>Manual</th>
<th>Relay</th>
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</thead>
<tbody>
<tr>
<td>Works over ISP’s NATs?</td>
<td>N/A</td>
<td>Ltd. (*)</td>
<td>N/A</td>
<td>N/A</td>
<td>Maybe</td>
</tr>
<tr>
<td>Symmetric NATs?</td>
<td>N/A</td>
<td>No</td>
<td>N/A</td>
<td>ok</td>
<td>Ltd.</td>
</tr>
<tr>
<td>Phone support needed?</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>NAT support needed?</td>
<td>Yes</td>
<td>Ltd. (*)</td>
<td>Yes</td>
<td>Ltd. (+)</td>
<td>No</td>
</tr>
<tr>
<td>Scalability</td>
<td>? (o)</td>
<td>Ok</td>
<td>Ok</td>
<td>Ok</td>
<td>poor ❌</td>
</tr>
<tr>
<td>User Effort</td>
<td>Small</td>
<td>Small</td>
<td>Small</td>
<td>Big ❌</td>
<td>Small</td>
</tr>
</tbody>
</table>

* … does not work for symmetric NATs  
  o … application-awareness affects scalability  
  + … port translation must be configurable

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NAT Traversal Scenarios

• There is no “one size fits it all” solution. All current practices suffer from many limitations.

• iptel.org observations for residential users behind NATs: Affordability wins: SIP-aware users relying on public SIP server use ALGs or STUN. First UPnP uses sighted.

• Our plan: hope for wider deployment of
  – STUN and STUN-friendly firewalls
  – ALGs
  – UPnP-enabled phones and NATs
Murphy’s Law Holds

*Everything can go wrong.*

- **Servers:**
  - software/configuration upgrades
  - vulnerabilities
  - both SIP and supporting servers subject to failure: DNS, IP routing daemons

- **Hosts:**
  - power failures
  - hard-disk failures

- **Networks:**
  - line.
  - IP access
IP Availability: SLAs

- Industry averages for “Network Availability” SLAs are from 99.9% to 99.5% (an NRIC report)
- SLAs mostly exclude regular maintenance and always Acts of God
- Residential IP access rarely with SLAs

<table>
<thead>
<tr>
<th>Availability (percent)</th>
<th>Actual Downtime (per year)</th>
</tr>
</thead>
<tbody>
<tr>
<td>99.999</td>
<td>5 Minutes</td>
</tr>
<tr>
<td>99.9</td>
<td>9 Hours</td>
</tr>
<tr>
<td>99.5</td>
<td>1.8 Days</td>
</tr>
</tbody>
</table>
matrix.net’s Reachability Statistics

- Minimum 98.69%
- Median 99.45%
- Maximum 99.84%
- Mean 99.40%

Wenyu Jang, Henning Schulzrinne: “Assessment of VoIP Service Availability in the Current Internet”, in PAM 2003. … 99.5%
Fail-over Issues

• Whatever the reason for a failure is, signaling needs to be available continuously. Most important components are:

• Replication of user information
  – Doable; using SIP gains better interoperability and avoids issues with database caches.

• Making clients use backup infrastructure on failure
  – SIP specification can do that (DNS/SRV) but today’s SIP phones cannot (except one).
Fail-over Workarounds and Limitations

• IP Address Take-over: Make backup server grab primary’s IP address when a failure detected
  – Cannot be geographically dispersed, unless coupled with re-routing
  – Primary server needs to be disconnected
• DNS Update: Update server’s name with backup’s IP Address
  – DNS propagation may take too long, even if TTL=0 (which puts higher burden on clients)
• Both methods rely on error detection which may be tricky – a pinging host may be distant from another client and have a different experience
Scalability Concerns

• New applications, like presence, are very talkative
  – Presence status update frequent
  – Each update ventilated to multiple parties

• Broken or misconfigured devices account for a fair load share; few of many real-world observations:
  – Broken digest clients resend wrong credentials in an infinite loop $\rightarrow$ heavy flood
  – Mis-configured password: a phone attempted to re-register every ten minutes (factor 6) $\rightarrow$ 2400 messages a day
  – Mis-configured Expires=30 (factor 120)

• Replication, Boot avalanches, NAT refreshes

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Achievable Scalability

- Good news: well-designed SIP servers can cope with load in terms of thousands of calls per second (CPS)
  - Example: lab-tuned version of SIP Express Router achieved transactional throughput in thousands of Calls Per Second on a dual-CPU PC – capacity needed by telephony signaling of Bay Area

- Pending concern: denial of service attacks
  - Example: hundreds of megabytes of RAM can be exhausted in tens of seconds with statefull processing

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Deployability

• Devices can be made scale, administrators not
• Well-known burdens:
  – Many boxes deployed consume many administrators.
    • Network-building practice: Integrate signaling logic in as few boxes as redundancy strategy allows.
  – Phones are not yet plug-and-play, particularly if behind NATs
    • It is still phone vendors’ turn.
  – SIP routing good but not easy (…)

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SIP Routing

- Benefit of SIP: Ability to link various service components together.
- The “glue” are signaling servers. Their primary capability is routing requests to appropriate services.

- Issues:
  - Routing flexibility – how to determine right destination for a request
  - Troubleshooting when routing failures occur

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Routing Was Never Easy

- Request processing policy may be quite complex:
  - PSTN destinations require SIP servers to stay in the path for purpose of accounting and admission control.
  - Some destinations are reachable for anonymous callers whereas others take authentication and admission control.
  - Requests from originators known to support NAT traversal may receive different treatment.
  - Method-based routing – requests to PSTN are split by method between SMS and PSTN gateway.
  - Further factors include request’s transport origin, address claimed in From header field, content of Contact, etc.

- Operational observation: mighty tools for specification of routing policy are needed.
Routing Language

• Our answer: routing language
• Features: conditional expressions may depend on any of previously mentioned factors; example:

```c
/* free destinations, like Jiri’s mobile phone listed in an SQL table, or any local PBX numbers require no authentication */
if ( is_user_in("Request-URI", "free-pstn") | uri=~"sip:[79][0-9][0-9][0-9][0-9]@.*" ) {
    log ("free call"); /* no admission control – let anyone call … */
} else { /* all other destinations require proper credentials */
    if (!proxy_authorize("iptel.org" /* realm */,"subscriber" /* table name */) {
        proxy_challenge("iptel.org", 0);
        break;
    }
    /* detailed admission control – long distance versus international, etc…*/
    if (uri=~"sip:0[1-9][0-9]+@.*") {
        if (!is_in_group("local")) {
            sl_send_reply("403", "Forbidden...");
        }
    }
...
```
SIP Routing: Troubleshooting

- SIP request can be routed along arbitrarily complex path
- Failures in numbering plans and SIP-routing in general difficult to locate without knowledge of:
  - Which Request URI caused an error
  - At which spiral iteration an error occurred
  - Who was the pre-last hop
  - Who was the next-hop when forwarding failed
Troubleshooting Proposal

• Operators do not know what is going wrong:
  – servers causing an error located on CP or belonging to a different administrative domain
  – users cannot report to operator on what is happening
• Proposal: take a lesson from email and include original message in replies – it includes all one needs to know
• Status: Already deployed at iptel.org, automated troubleshooting would take standardization
Concluding Observations

- Basic VoIP & complementary services up and running.
- Performance essential to survival of critical situations such as mis-configured networks and to avoidance of too many servers. Denial of Service still a pending challenge.
- Request-routing flexibility in servers essential to building services, but it takes troubleshooting facilities.
- Improvement place for phone implementations still exists: NAT traversal support, plug-and-play configuration, DNS fail-over.
Information Resources

• Email: jiri@iptel.org
• IP Telephony Information: http://www.iptel.org/info/
• SIP Services: http://www.iptel.org/user/
• SIP Express Router: http://www.iptel.org/ser/
• Related RFCs and Internet Drafts: 
  http://www.iptel.org/info/
  • NATs: draft-ietf-sipping-nat-scenarios-00.txt
  • Diagnostic: draft-kuthan-sipping-diag-00.txt