Overview

- Introduction to SIP
- Addresses and Address Resolution in SIP
- ENUM & SIP
- Peer-to-Peer for SIP Telephony
- Conclusion
IETF Conferencing

Session Initiation Protocol (SIP, RFC 3261)

- Initiate, terminate, and modify sessions
  - Multimedia(!) sessions (not just voice!)
  - Point-to-point and multiparty

- Support for
  - caller and callee authentication / call authorization
  - privacy for call signaling and media streams
  - media path with ensured QoS
  - policy-based control mechanisms

- Flexible service creation
  - end-to-end principle (“dumb network”)
  - support through SIP servers (located anywhere)

- Extensible protocol to cover new communication aspects
  - such as presence and instant messaging
Terminology

- SIP User Agent
  - User Agent Server (UAS)
  - User Agent Client (UAC)

- SIP Registrar
- SIP Location Server

- SIP Redirect Server
- SIP Proxy Server
- SIP Back-to-Back (B2B) User Agent

- SIP Application Servers

Local SIP Architecture

Administrative Entity (SIP Server)
- Registrar
- Redirect / Proxy Server
- Location Server

Local IP network
- Endpoint
  - SIP UA
- Endpoint
  - SIP UA
- Endpoint
  - SIP UA

SIP Gateway
- SIP UA
- SIP UA
- SIP UA

PSTN
ISDN

GSM

H.323
Naming: SIP URIs

- Separating names (permanent) and addresses (temporary)
  - Basic mobility support

- Two roles reflected in SIP
  - Naming a user; typically `sip:user@domain`
  - Contact address of a user; typically contains host name or IP address, port, transport protocol, ...
  - May also refer to a domain as a whole

- URIs may carry additional parameters

- URIs may also identify services

SIP URI Addressing Examples

```
sip:tzi.org
```
Registration domain or IP address

```
sips:192.168.42.1
```
SIP URI to call (AoR)

```
sip:john@example.com
```
SIP address of actual user location

```
sip:john@host1.example.com
sip:john@192.168.42.9:9950
```

```
sip:voicemail@service.com
sip:conf-1234@confserv.com
sips:user34@anonymizer.org
```
Service identifier; semantics opaque to the user
Further Common URI Schemes

Telephony (RFC 3966)
- tel:+1-555-12345678
- tel:7595;phone-context=+49421218

ITU-T H.323 Protocol
- h323:user@example.com

Instant Messaging
- im:user@example.com

Presence
- pres:user@example.com

SIP Message Syntax: Request

Start line
```
INVITE sip:user@example.com SIP/2.0
```
To: John Doe <sip:user@example.com>
From: sip:jo@tzi.uni-bremen.de;tag=4711
Subject: Congratulations!
Content-Length: 117
Content-Type: application/sdp
Call-ID: 2342344233@134.102.218.1
CSeq: 49581 INVITE
Contact: sip:jo@134.102.224.152:5083;transport=udp
Via: SIP/2.0/UDP 134.102.218.1

Message body (SDP content)
```
v=0
o=jo 75638353 98543585 IN IP4 134.102.218.1
s=SIP call
t=0 0
c=IN IP4 134.102.224.152
m=audio 47654 RTP/AVP 0 1 4
```
SIP Message Syntax: Response

Start line

```plaintext
200 OK SIP/2.0
```

Message headers

- To: John Doe <sip:user@example.com>;tag=428
- From: sip:jo@tzi.uni-bremen.de;tag=4711
- Subject: Congratulations!
- Content-Length: 121
- Content-Type: application/sdp
- Call-ID: 2342344233@134.102.218.1
- CSeq: 49581 INVITE
- Contact: sip:jdoe@somehost.domain
- Via: SIP/2.0/UDP 134.102.218.1

Message body (SDP content)

```plaintext
v=0
o=jdoe 28342 98543601 IN IP4 134.102.20.22
s=SIP call
t=0 0
c=IN IP4 134.102.20.38
m=audio 61002 RTP/AVP 0 4
```

Example: Direct Call UA–UA
Direct Call

**INVITE**

```
sip:bob@foo.bar.com
```

**Note:** Three-way handshake is performed only for INVITE requests.

**Media Streams**

Call established by Bob acknowledging the INVITE request:

- Media data are exchanged (e.g., RTP)
- Call is terminated by one participant

How to Find The Callee?

- Direct calls require knowledge of callee’s address
- SIP provides abstract naming scheme:
  
  ```
sip:user@domain
  ```

  → Define mapping from SIP URI to real locations:
     - Explicit registration:
       UA registers user’s name and current location
     - Location service:
       Use other protocols to find potentially correct addresses
- Caller sends INVITE to any SIP server knowing about the callee’s location
- Receiving server may either redirect, refuse or proxy
Finding the Next Hop

- UAC may use a (manually) configured outbound proxy
  - Outbound proxy may also have be learned upon registration

- If request URI contains IP address and port, message can be sent directly

- Otherwise, determine next hop SIP server name via DNS
  - Use NAPTR RR (SIP+D2U/D2T/D2S, SIPS+D2T/D2S) to obtain SRV records
  - Query for SRV RR: _sip._udp, _sip._tcp, _sips._tcp for all supported transport protocols
  - If entries found, try as specified in RFC 2782
    - If no entries found, use UDP for sip: URIs and TCP for sips: URIs

- Query A or AAAA records for IP address
  - For specified domain name
  - (Deprecated: For specified sip.domain )
Proxied Call

INVITE sip:bob@bar.com
100 Trying
200 OK
ACK

INVITE sip:bob@foo.bar.com
100 Trying
200 OK

Media Streams
Subsequent requests

Global SIP Architecture

SIP signaling for initial call routing and setup
SIP in-call signaling
RTP media streams
SIP Address Resolution Steps

1. Client uses configuration information to locate outbound proxy. May use DNS for address resolution. All requests sent to proxy.

sip:ob@tzi.org  sip:jo@acm.org
SIP Address Resolution Steps

1. Proxy checks for domain suffix. Consults location service (e.g., DNS).
2. Uses DNS to locate remote domain. DNS yields address of proxy 2.
3. Proxy checks for domain suffix. Consults location service (registration) to determine current address of jo@acm.org.
SIP Address Resolution Steps

1. SIP UA 1 requests a call from sip:ob@tzi.org to sip:jo@acm.org.
2. The request is routed through SIP Proxy 1 to SIP Proxy 2.
3. SIP Proxy 2 checks the Domain Name System (DNS) for sip:jo@acm.org.
4. UA 2 validates the SIP URI against active users to alert the right one or reject.

Address Resolution for Phone Numbers

1. SIP UA 1 requests a call from sip:+49-421-218-1 to tel:+49-421-201-7028.
2. The request is routed through SIP Proxy 1 to SIP Proxy 2.
3. SIP Proxy 2 checks the Domain Name System (DNS) for sip:+49-421-201-7028.
Finding the Next Hop for tel: URIs

- **UAC may use a (manually) configured outbound proxy**
  - Outbound proxy may also have be learned upon registration

- If request URI contains IP address and port, message can be sent directly

- Otherwise, **determine next hop SIP server via DNS**
  - Use NAPTR RR (SIP+D2U/D2T/D2S, SIPS+D2T/D2S) to obtain SRV records
  - Query for SRV RR: _sip._udp, _sip._tcp, _sips._tcp for all supported transport protocols
  - If entries found, try as specified in RFC 2782
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- **Query A or AAAA records for IP address**

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**ENUM Process**

- ENUM service lookup
- Use DNS NAPTR Resource Records

- NAPTR RR lookup to select preferred SIP services
- Based upon transport protocols and TLS

- SRV RR lookup for load balancing
  - May or may not yield IP address

- A or AAAA lookup to determine IP address(es) associated with name

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Example:

- tel:+49-421-201-7028
- sip:jo@tzi.org
- mailto:jo@tzi.org
- _sip._tcp.tzi.org: jo@_sip._udp.tzi.org
- sip:jo@damn.tzi.org: 5060;transport=tcp
- sip:jo@134.102.218.67: 40987;transport=tcp
Background

- Background: Dynamic Delegation Discovery System (DDDS)
  - RFC 3401, 3402, 3403, RFC 3761, RFC 3764

- Abstract concept
  - Map Application Unique String (AUS) to some data
  - Based upon a (distributed) database
  - Indexed by keys
  - Lookup yields rule set: matching + substitution rules (regexp-like)
  - Result may point to a different location in the database
  - Algorithm: repeated substitution applied to AUS until a terminal symbol is reached

- Effect: Allow delegation of responsibility to store “some data” for an AUS

Step 1: DDDS General Operation

1. Application Unique String
2. First Well-Known Rule
3. First key
4. Identity
5. Key
6. Lookup key in database
7. AUS
8. Rule set
9. Key
10. Delegation of Responsibility
11. Desired result
12. Terminal rule?
13. Apply rule set to AUS until non-empty result is obtained that meets the application requirements
Step 1: NAPTR Resource Records

- Uses DDDS with DNS as distributed database
  - Key: DNS name

<table>
<thead>
<tr>
<th>Order</th>
<th>Preference</th>
<th>Flags</th>
<th>Services</th>
<th>regexp</th>
<th>Replacement</th>
</tr>
</thead>
</table>

- Order: absolute sorting of rules; first matching one is used
- Preference: application-specific priority indication
- Flags: control rule processing; e.g. indicate terminal rule
- Services: available services on this delegation path

- regexp: substitution rule for Application Unique String
  - Mutually exclusive
- Replacement: replacement for DNS name (= key)

Step 1: ENUM Lookup

- Application unique string: tel:+49-421-201-7028
  - +49-421-201-7028 ➔ +494212017028
- First well-known rule: identity
- Database key: transformation into valid DNS name
  1. Remove leading ‘+’: 494212017028
  2. Put dots between digits: 4.9.4.2.1.2.0.1.7.0.2.8
  3. Reverse order of digits: 8.2.0.7.1.0.2.1.2.4.9.4
  4. Add ‘e164.arpa’: 8.2.0.7.1.0.2.1.2.4.9.4.e164.arpa

- Yields the domain name used to query for NAPTR records

- Flags: ‘u’ to indicate terminal rule
- Service: E2U+servicespec[+servicespec]…
- ENUM services: sip, h323, pres, …
### Step 1: SIP Lookup for ENUM

<table>
<thead>
<tr>
<th>Type</th>
<th>Order</th>
<th>Preference</th>
<th>Flags</th>
<th>Services</th>
<th>regexp</th>
<th>Replacement</th>
</tr>
</thead>
<tbody>
<tr>
<td>NAPTR</td>
<td>10</td>
<td>100</td>
<td>u</td>
<td>E2U+sip</td>
<td>!^.*$!sip:<a href="mailto:jo@tzi.org">jo@tzi.org</a>!</td>
<td>.</td>
</tr>
<tr>
<td>NAPTR</td>
<td>10</td>
<td>101</td>
<td>u</td>
<td>E2U+pres</td>
<td>!^.*$!pres:<a href="mailto:jo@tzi.org">jo@tzi.org</a>!</td>
<td>.</td>
</tr>
<tr>
<td>NAPTR</td>
<td>10</td>
<td>102</td>
<td>u</td>
<td>E2U+msg</td>
<td>!^.*$<a href="mailto:jo@tzi.org">mailto:jo@tzi.org</a>!</td>
<td>.</td>
</tr>
</tbody>
</table>

#### Example: SIP (and other) entries for a user

$\textit{ORIGIN} 8.2.0.7.1.0.2.1.2.4.9.4.e164.arpa

**Mapping to a Domain Name**

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### Step 2: SIP NAPTR for Transport Selection

<table>
<thead>
<tr>
<th>Type</th>
<th>Order</th>
<th>Preference</th>
<th>Flags</th>
<th>Services</th>
<th>regexp</th>
<th>Replacement</th>
</tr>
</thead>
<tbody>
<tr>
<td>NAPTR</td>
<td>50</td>
<td>100</td>
<td>u</td>
<td>D2T+SIPS</td>
<td>_sips._tcp.tzi.org</td>
<td></td>
</tr>
<tr>
<td>NAPTR</td>
<td>90</td>
<td>100</td>
<td>u</td>
<td>D2T+SIP</td>
<td>_sip._tcp.tzi.org</td>
<td></td>
</tr>
<tr>
<td>NAPTR</td>
<td>100</td>
<td>100</td>
<td>u</td>
<td>D2U+SIP</td>
<td>_sip._udp.tzi.org</td>
<td></td>
</tr>
</tbody>
</table>

#### Example: Secure SIP preferred over SIP, TCP over UDP

$\textit{ORIGIN} tzi.org
Steps 3 and 4: SRV and A Resource Records

- Example: SIP load balancing across three servers

$ORIGIN _sip._tcp.tzi.org

<table>
<thead>
<tr>
<th>IN SRV</th>
<th>0</th>
<th>1</th>
<th>5060</th>
<th>damn.tzi.org</th>
</tr>
</thead>
<tbody>
<tr>
<td>IN SRV</td>
<td>0</td>
<td>2</td>
<td>5060</td>
<td>rasen.tzi.org</td>
</tr>
<tr>
<td>IN SRV</td>
<td>0</td>
<td>4</td>
<td>50600</td>
<td>rasen.tzi.org</td>
</tr>
</tbody>
</table>

- Finally: lookup of A records for rasen.tzi.org
- Then send SIP message to 134.102.218.67

Alternative Address Resolution Schemes

- Telephony Routing for IP (TRIP) [RFC 3219]
  - BGP-4-based routing protocol to find gateways

- Hierarchical Routing
  - See H.323 Gatekeeper Hierarchy across NRENs

- Static routing
  - SIP-based IP PBXes with statically configured prefix routing

- Peer-to-peer address resolution
  - Relying on a different distributed data base than DNS

...
Peer-to-Peer SIP

DNS $\rightarrow$ P2P: Distributed Hash Table System

SIP UA 1

sip:ob@tzi.org

SIP Proxy 1

SIP Proxy 2

SIP UA 2

sip:jo@tzi.org

Peer-to-Peer SIP: UAs as P2P Nodes

P2P: Distributed Hash Table System

SIP UA 1

sip:ob@tzi.org

SIP UA 2

sip:jo@tzi.org
Peer-to-Peer SIP: Proxies as P2P Nodes

P2P: Distributed Hash Table System

Two major issues: trust and reliability

sip:ob@tzi.org
sip:jo@tzi.org

Conclusion

- ENUM defines a way of resolving phone numbers to SIP entities
  - Supports first level of service identification
  - Makes use of DNS as a distributed database
  - Fits well with the regular SIP address resolution process

- Other address resolution protocols equally conceivable
Conclusion

- ENUM defines a way of resolving phone numbers to SIP servers
  - Supports first level of service identification
  - Makes use of DNS as a distributed database
  - Fits well with the regular SIP address resolution process

- Other address resolution protocols equally conceivable

Debate about the Future: [Henry Sinnreich]

- Traditional SIP for enterprise deployments
- Peer-to-peer for private users?
- Carriers?

ISOC.de und die IETF

- Und am Schluß: Werbung!

- Laufende Entwicklungen der IETF zu SIP, ENUM, …

- Breites Interesse in DE, dennoch begrenzte Teilnehmerzahl

- Idee: Mentoring für neue Interessenten
  - Hintergrundinformationen, Einführung, Ko-Autorenschaft bei Dokumenten


- Mail an ietf-tag@isoc.de