SIP and ENUM

2005-03-01 ENUM-Tag @ DENIC

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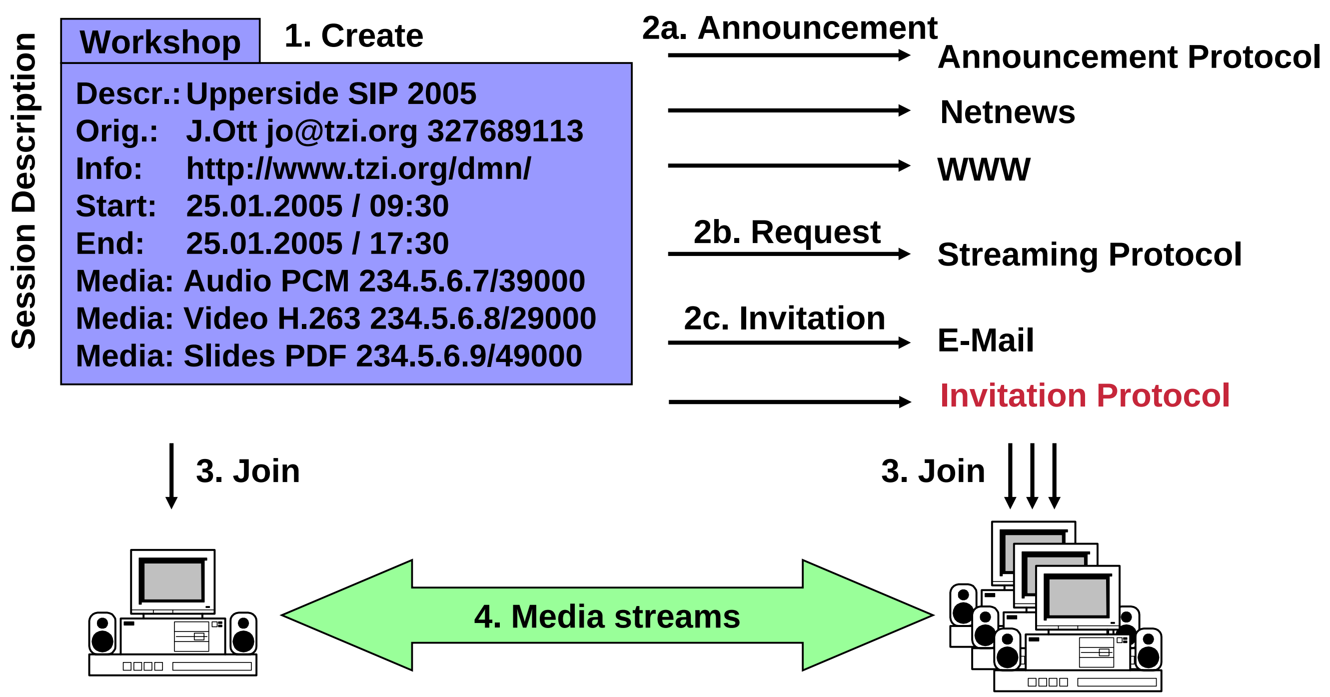
Overview

- 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)

User Agent = Endpoint, Gateway

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

IP Telephony Server

- SIP Application Servers

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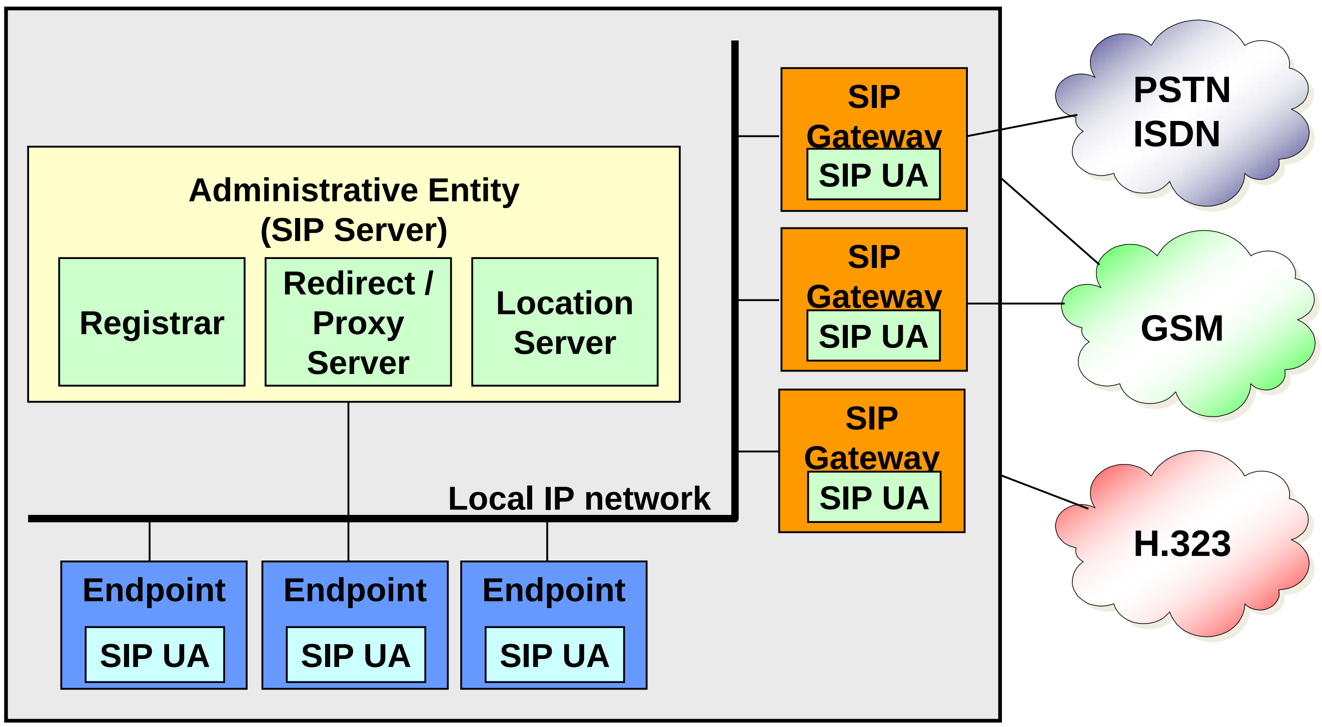
Local SIP Architecture



Naming: SIP URIs

sip: / sips:

- 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

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SIP URI Addressing Examples

`sip:tzi.org` `sips:192.168.42.1`	**Registration domain or IP address**
`sip:john@example.com`	**SIP URI to call (AoR)**
`sip:john@host1.example.com` `sip:john@192.168.42.9:9950`	**SIP address of actual user location**

`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

Message headers

Message body (SDP content)

```
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: applicaton/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

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

```
200 OK SIP/2.0
```

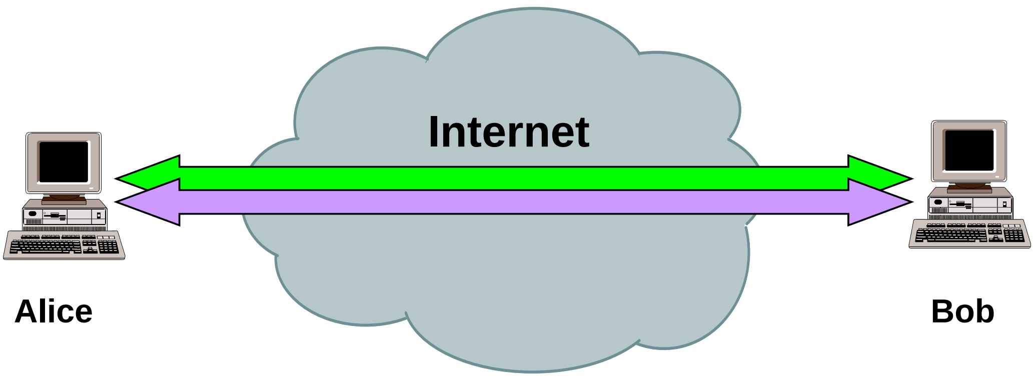
Message headers

```
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: applicaton/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)

```
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
```

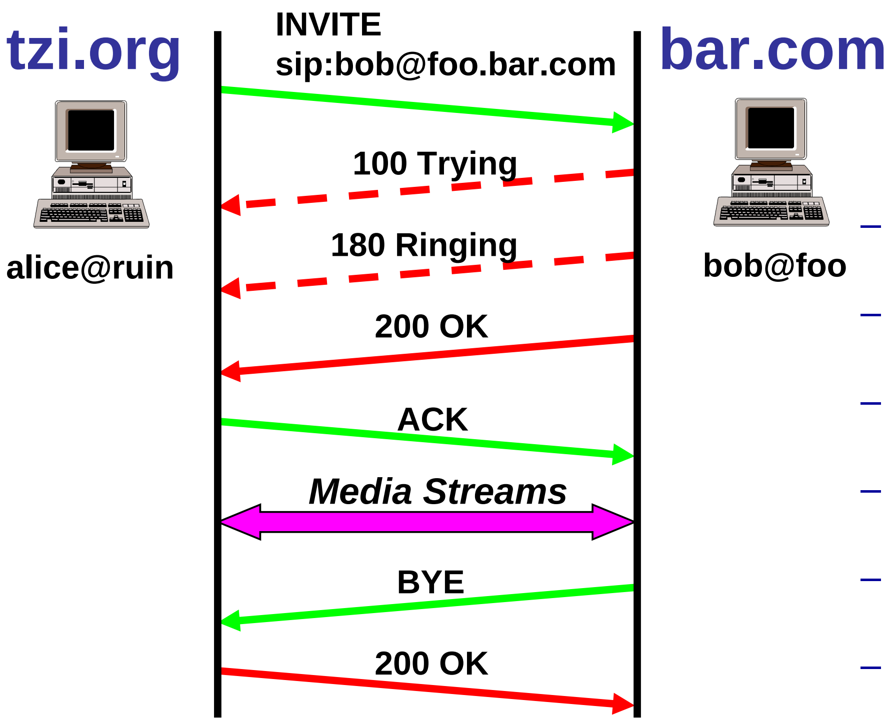

Example: Direct Call UA–UA



Call signaling

Media streams

Direct Call



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

- **Caller knows callee's hostname or address**
- Called UA reports status changes
- After Bob accepted the call, OK is signaled
- Calling UA acknowledges, call is established
- 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:

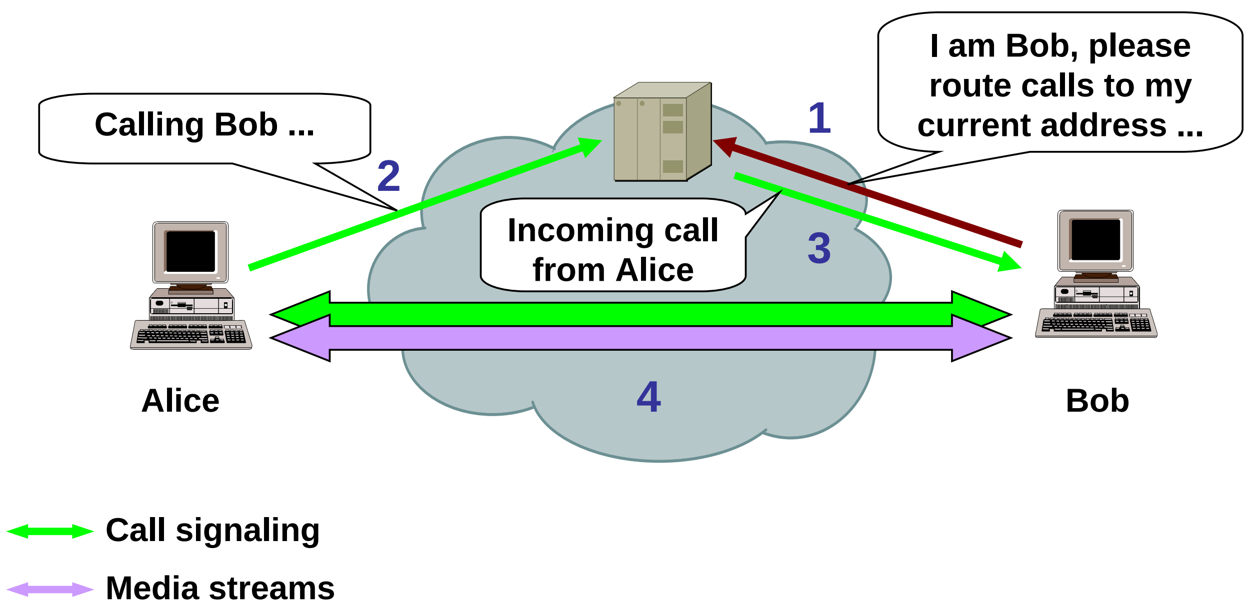
sip:user@domain

→ 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

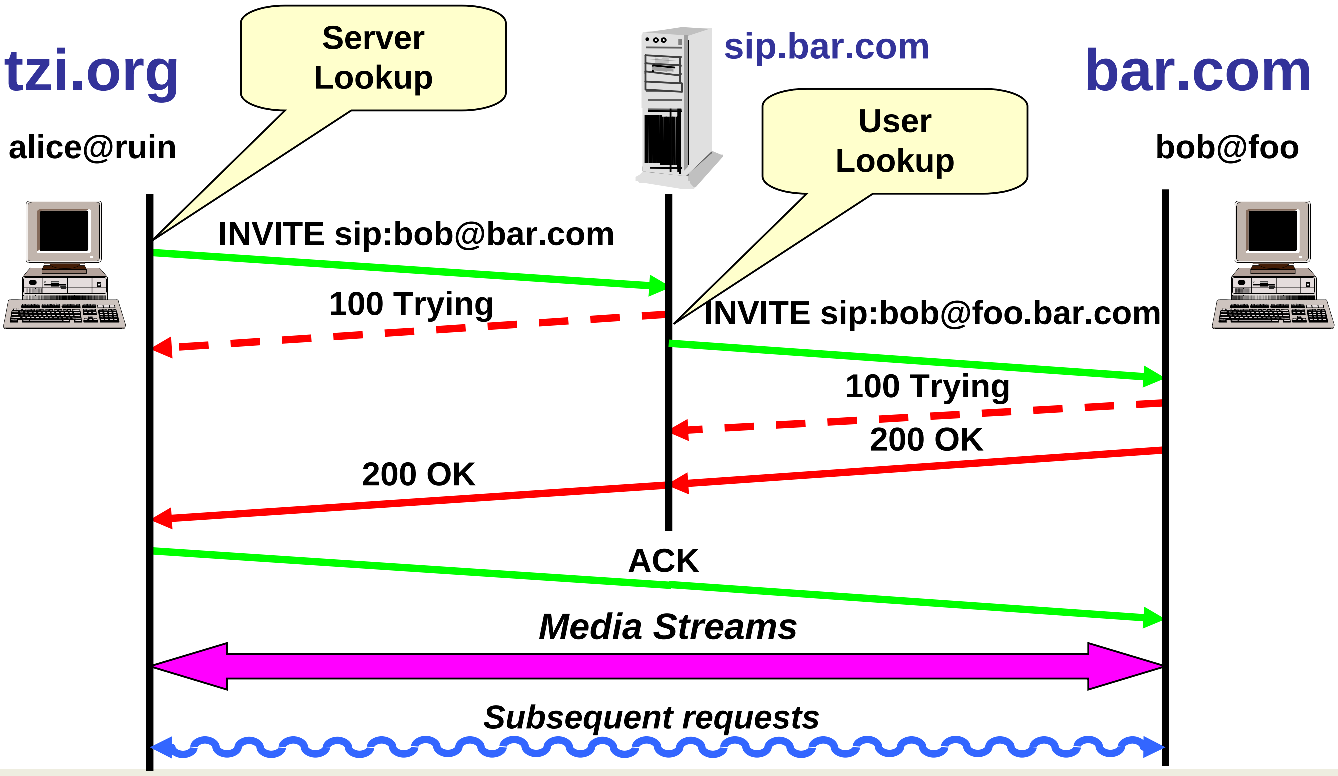
Finding the Next Hop

- **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*)

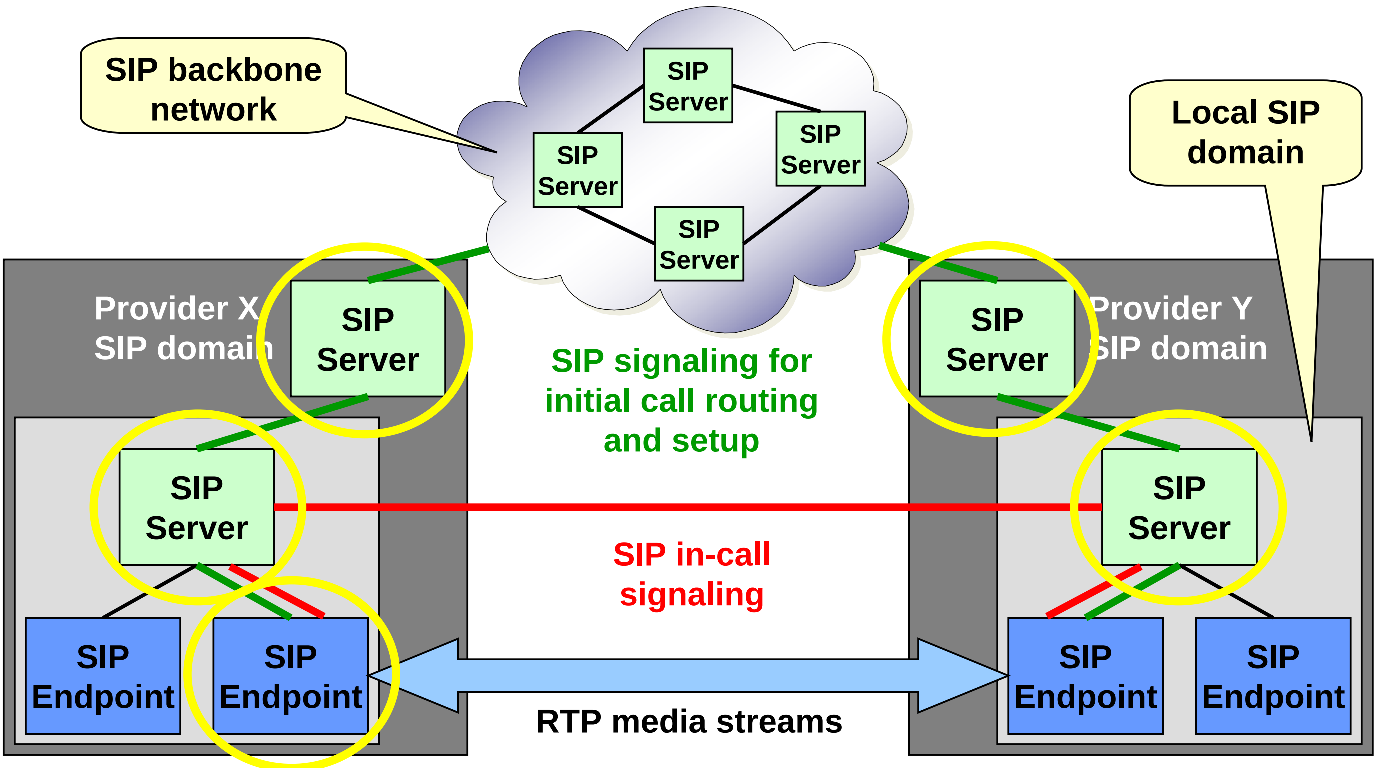
Simple Scenario of a SIP Call



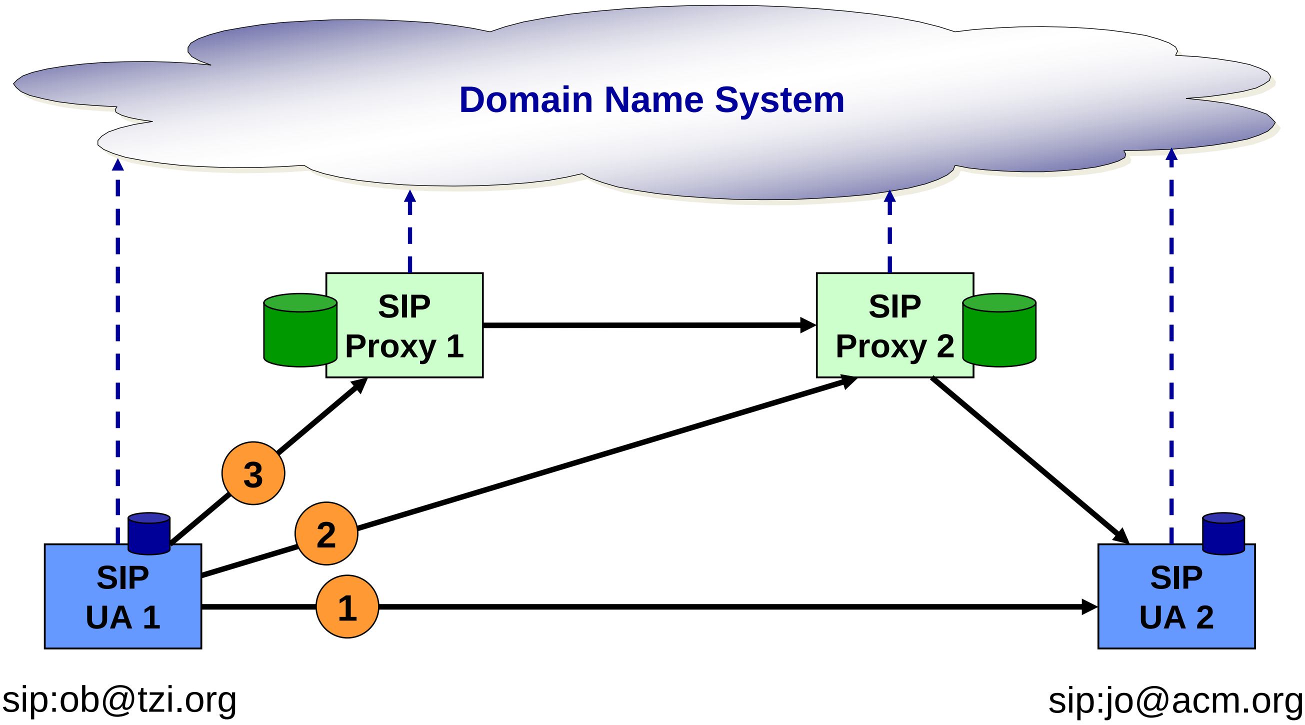
Proxied Call



Global SIP Architecture



SIP Address Resolution Steps



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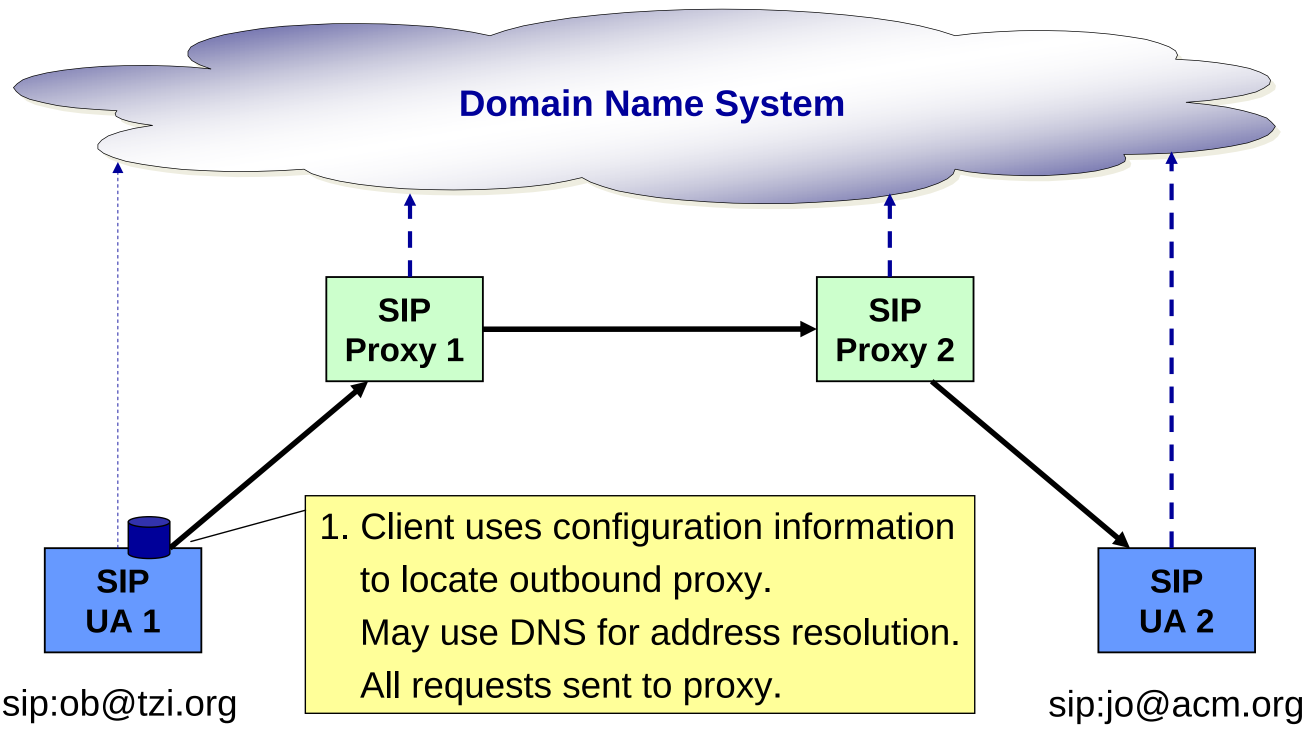

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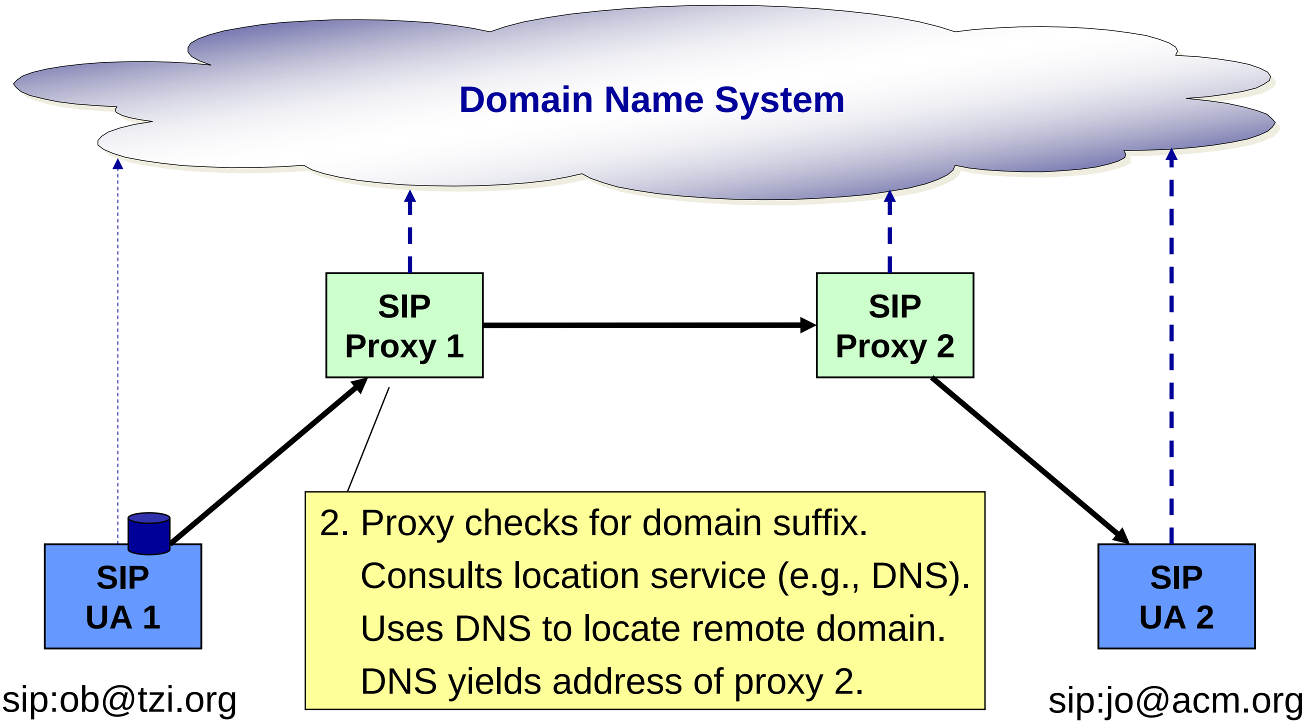

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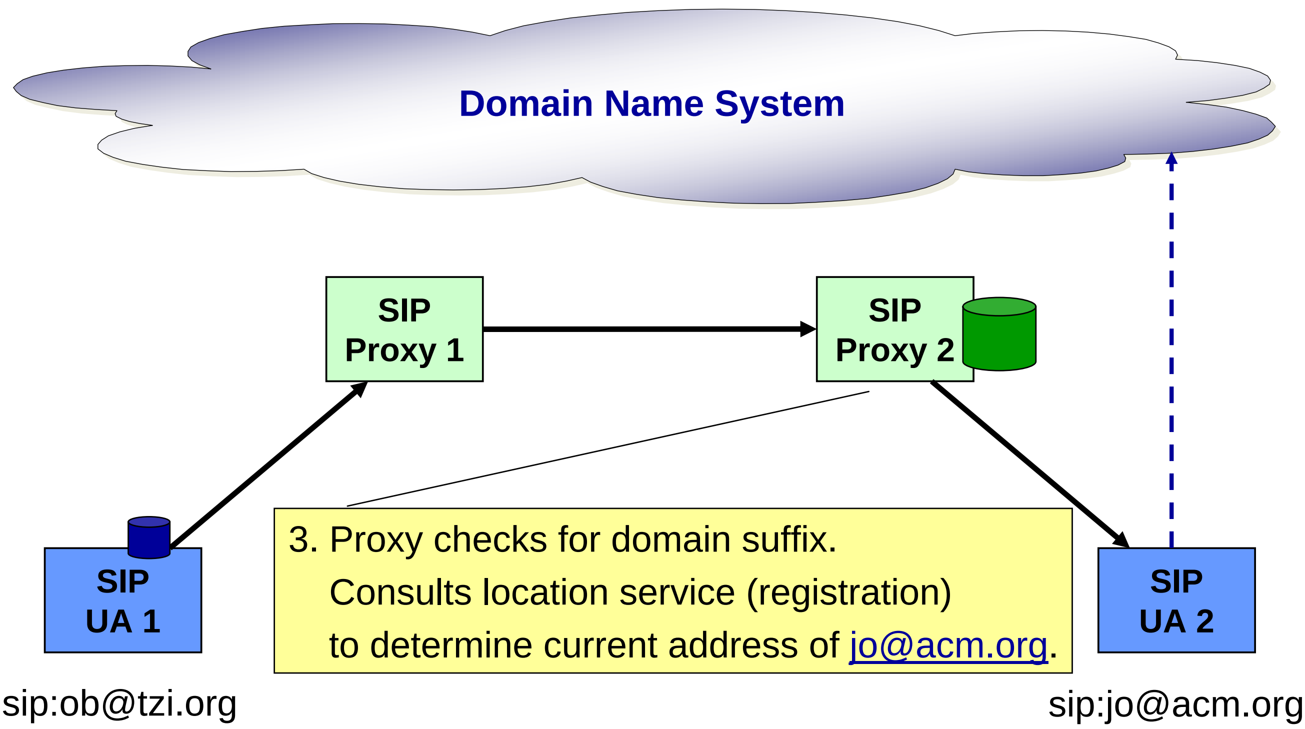

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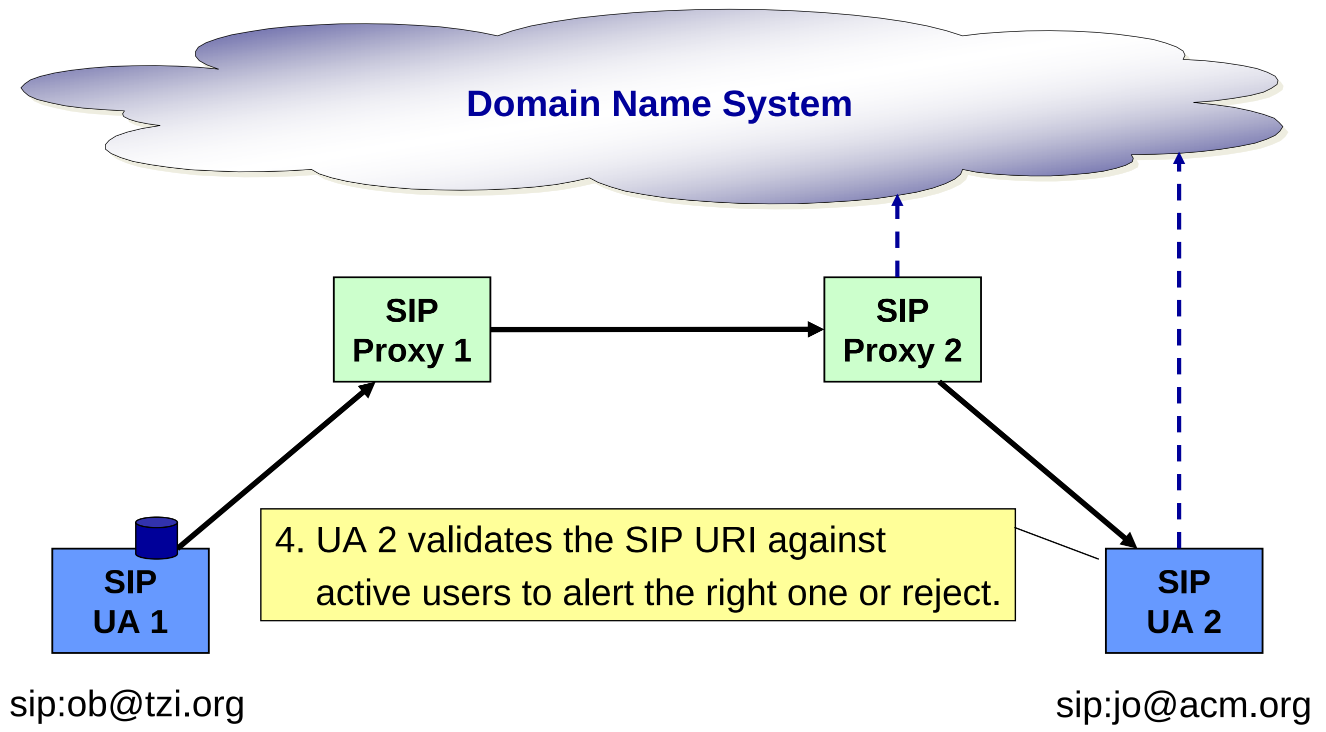
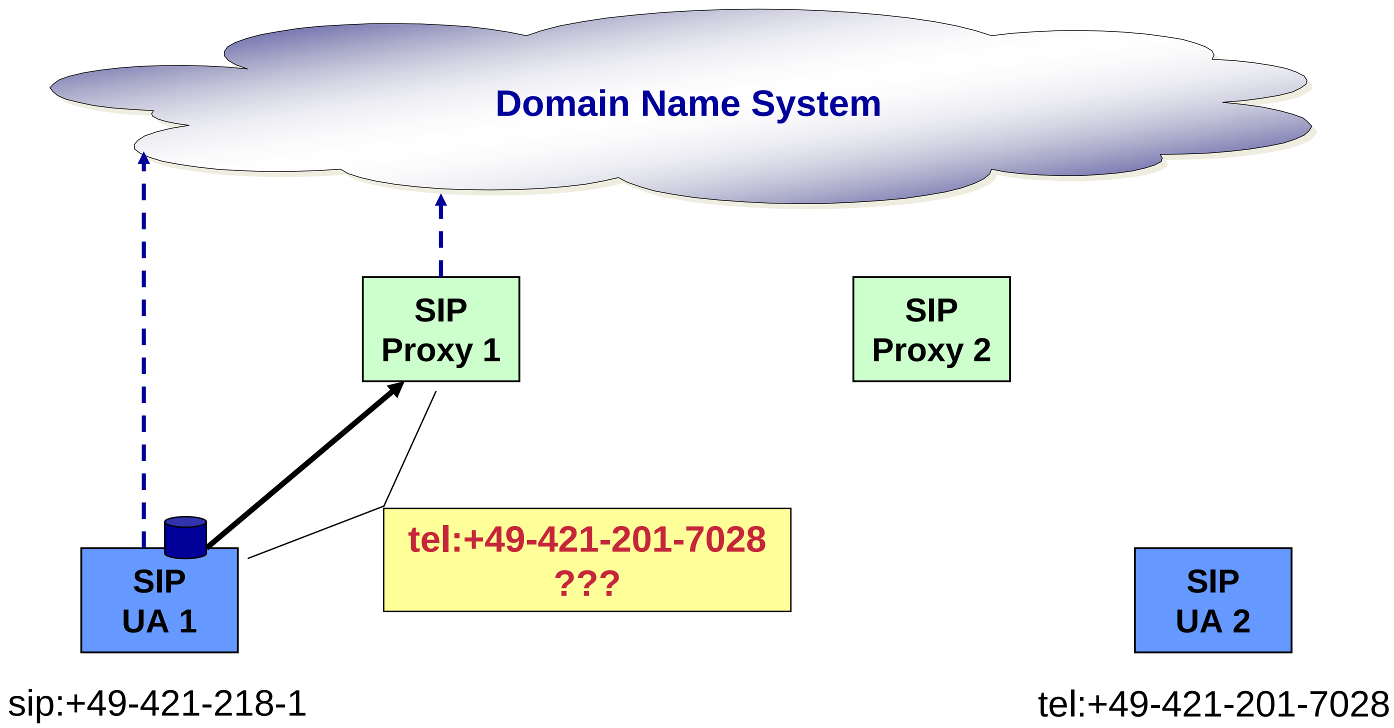


Address Resolution for Phone Numbers



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
    - If no entries found, use UDP for sip: URIs and TCP for sips: URIs
- **Query A or AAAA records for IP address**

ENUM Process

```
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
```

- ENUM service lookup
- Uses DNS NAPTR Resource Records

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

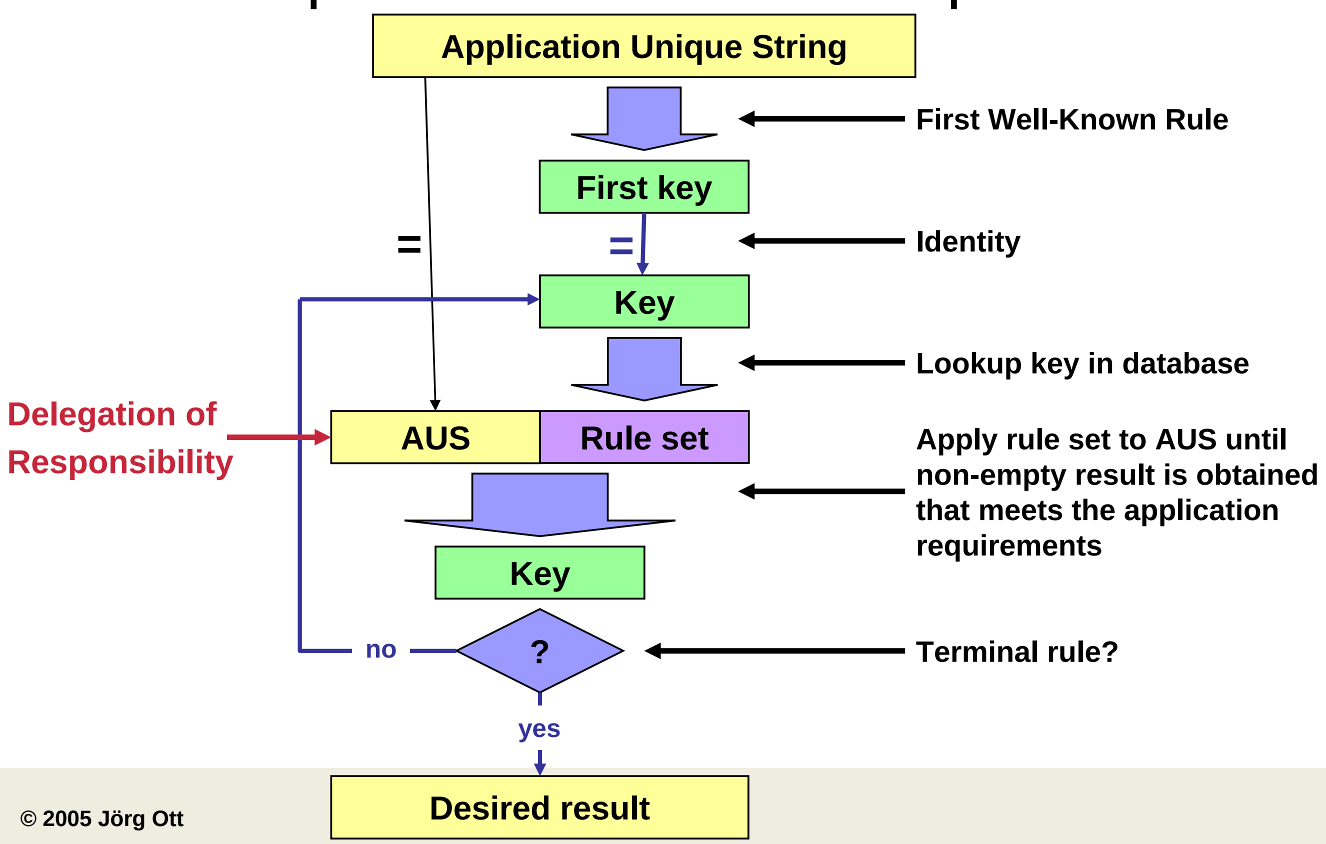
- SRV RR lookup for load balancing
- May or may not yield IP address

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

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



Step 1: NAPTR Resource Records

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

Order	Preference	Flags	Services	regexp	Replacement

- 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

Type	Order	Preference	Flags	Services	regexp	Replacement

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

$ORIGIN 8.2.0.7.1.0.2.1.2.4.9.4.e164.arpa

IN NAPTR	10	100	u	E2U+sip	!^.*$!sip:jo@tzi.org!	.
IN NAPTR	10	101	u	E2U+pres	!^.*$!pres:jo@tzi.org!	.
IN NAPTR	10	102	u	E2U+msg	!^.*$!mailto:jo@tzi.org!	.

Mapping to a Domain Name

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Step 2: SIP NAPTR for Transport Selection

Type	Order	Preference	Flags	Services	regexp	Replacement

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

$ORIGIN tzi.org

IN NAPTR	50	100	u	D2T+SIPS		_sips._tcp.tzi.org
IN NAPTR	90	100	u	D2T+SIP		_sip._tcp.tzi.org
IN NAPTR	100	100	u	D2U+SIP		_sip._udp.tzi.org

Steps 3 and 4: SRV and A Resource Records

- Example: SIP load balancing across three servers

$ORIGIN _sip._tcp.tzi.org

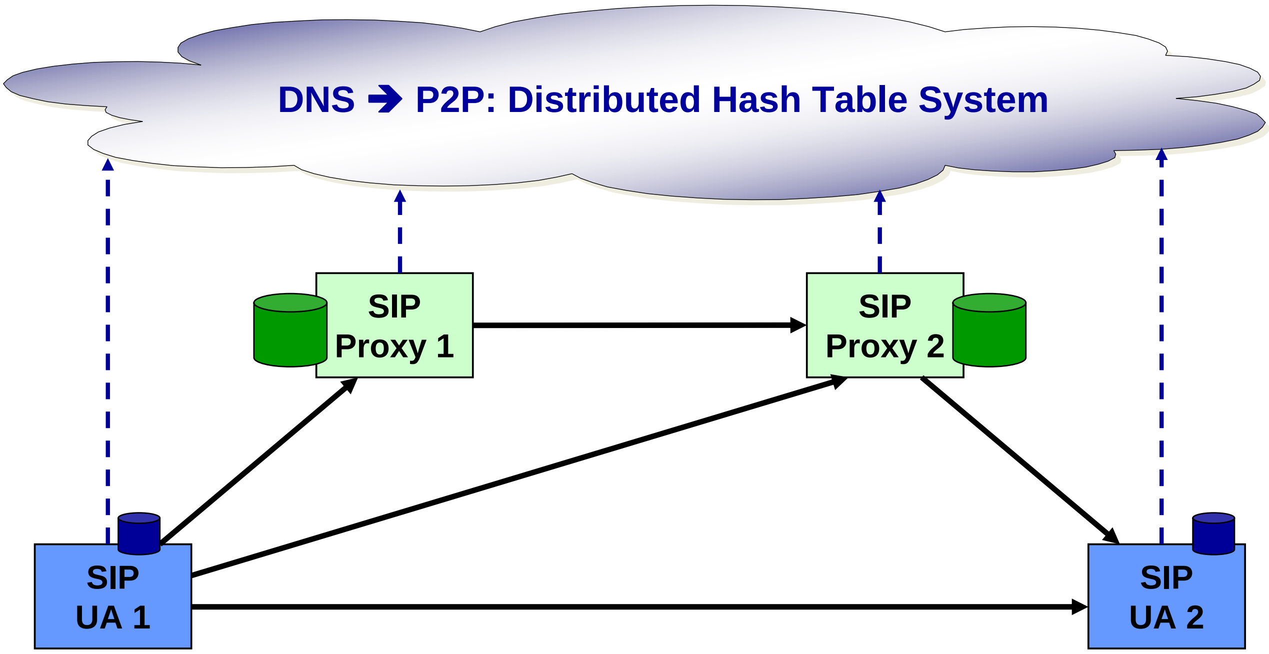
IN SRV	0	1	5060	damn.tzi.org
IN SRV	0	2	5060	rasen.tzi.org
IN SRV	0	4	50600	rasen.tzi.org

- 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
- …

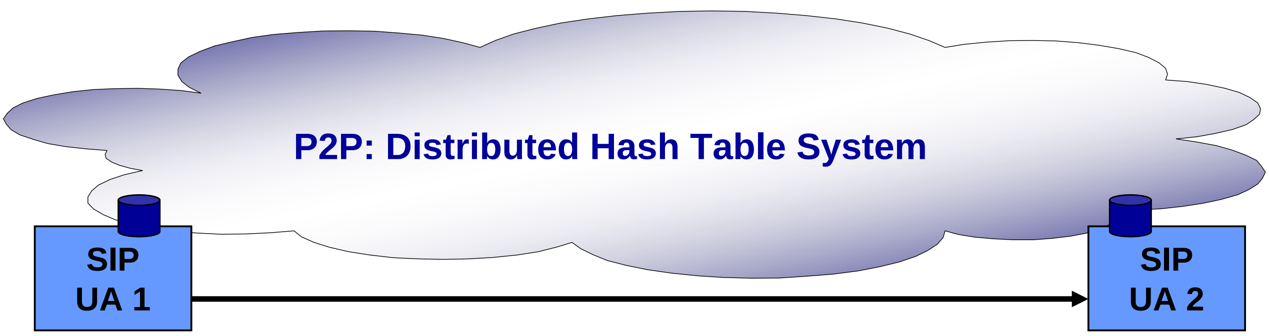
Peer-to-Peer SIP



sip:ob@tzi.org

sip:jo@tzi.org

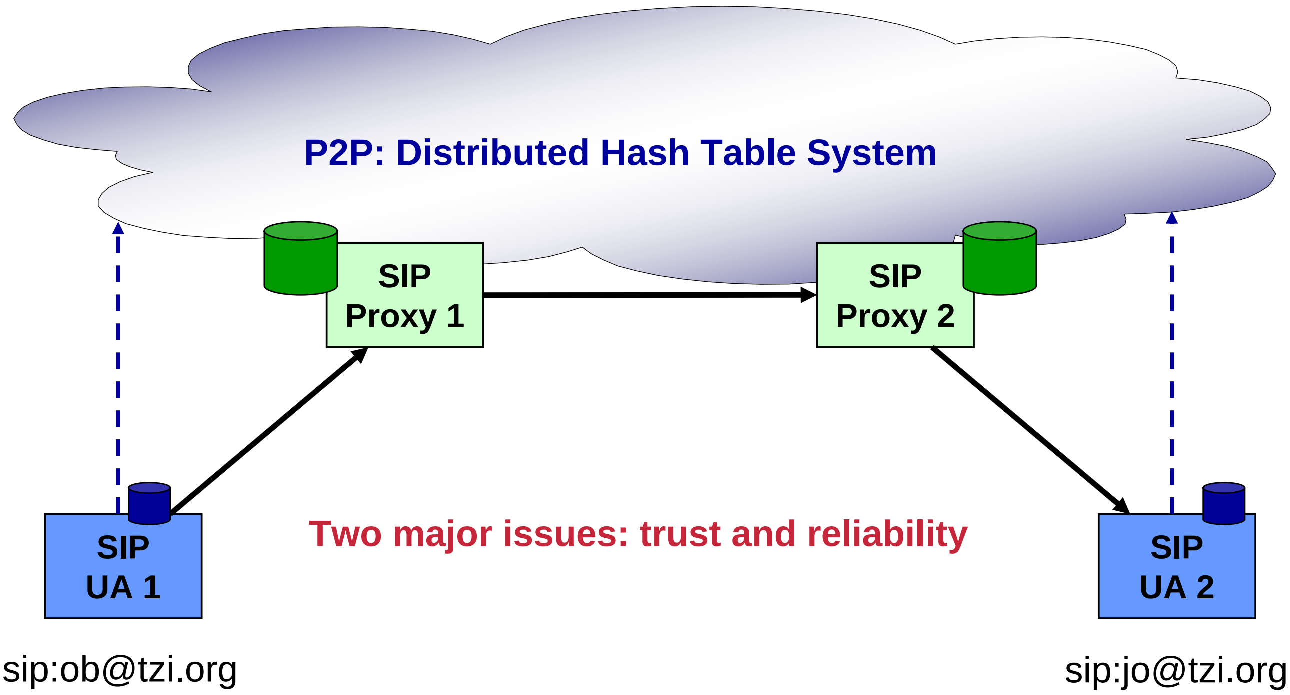
Peer-to-Peer SIP: UAs as P2P Nodes



sip:ob@tzi.org

sip:jo@tzi.org

Peer-to-Peer SIP: Proxies as P2P Nodes



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
  - IETF-Tag: 20./21.September 2005 Kassel-Wilhelmshöhe
- Nächste naheliegende Gelegenheit: Paris, 31.7. – 5.8.2005
- Mail an ietf-tag@isoc.de