



SIP and ENUM

2005-03-01

ENUM-Tag @ DENIC

Jörg Ott

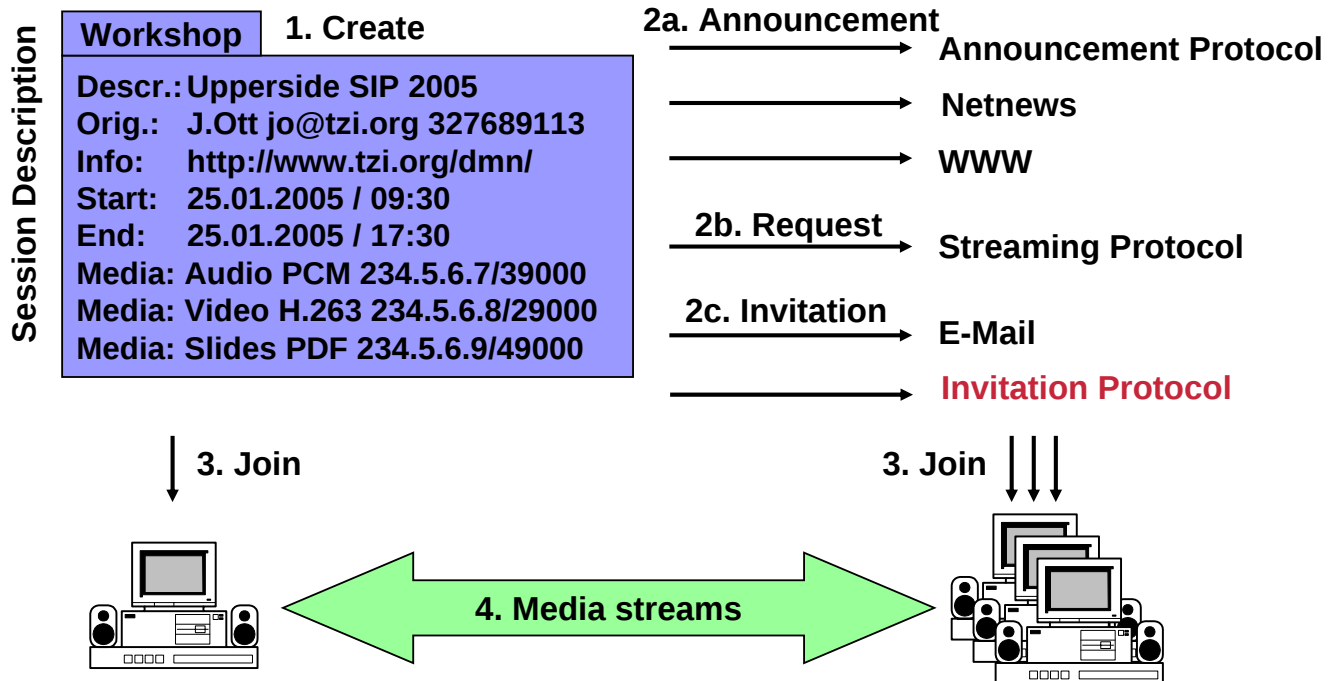
<jo@netlab.hut.fi>



Overview

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

IETF Conferencing



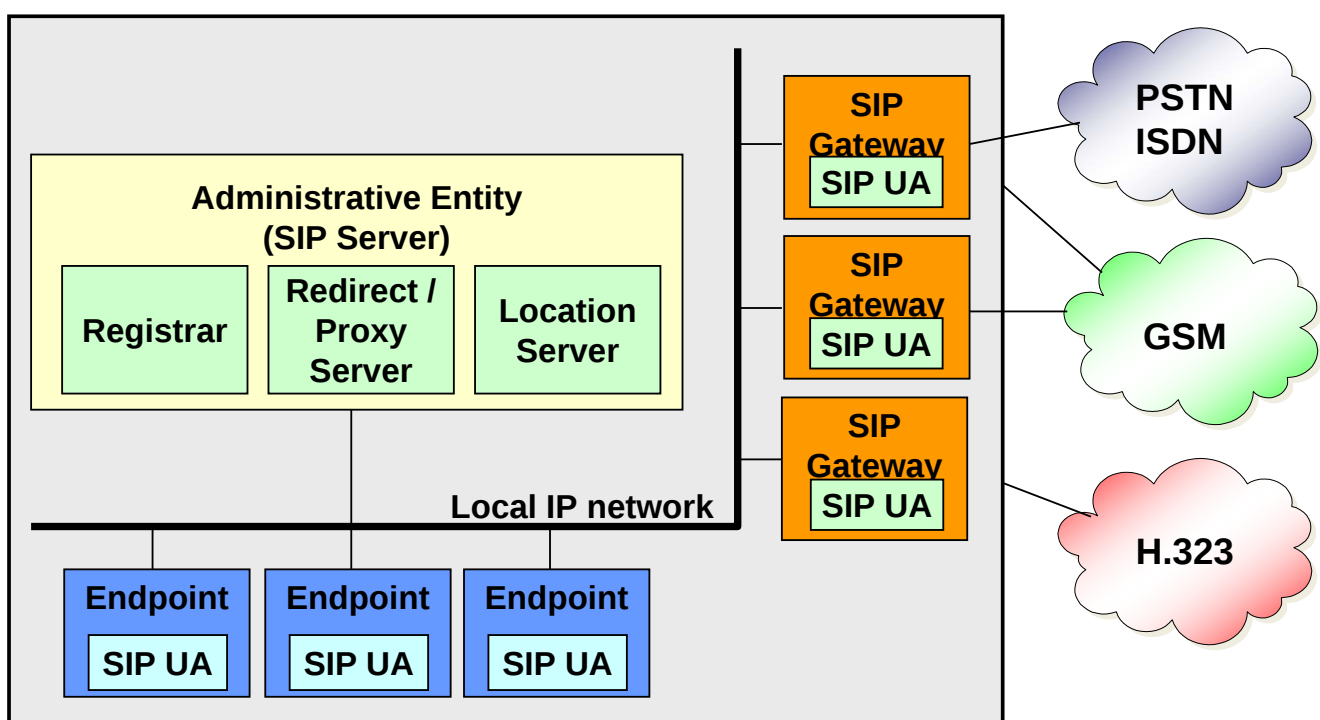
Session Initiation Protocol (SIP, RFC 3261)

- ▶ 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
- User Agent = Endpoint, Gateway**
- IP Telephony Server**

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

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

INVITE sip:user@example.com SIP/2.0

Message headers

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

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

200 OK SIP/2.0

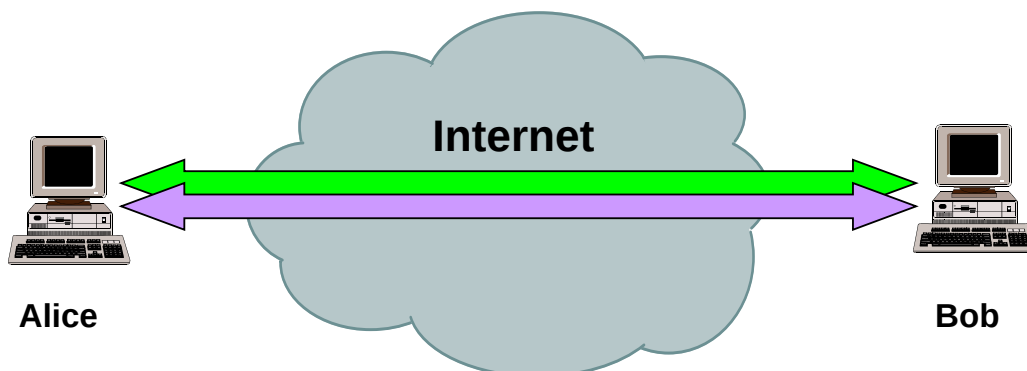
Message headers

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)

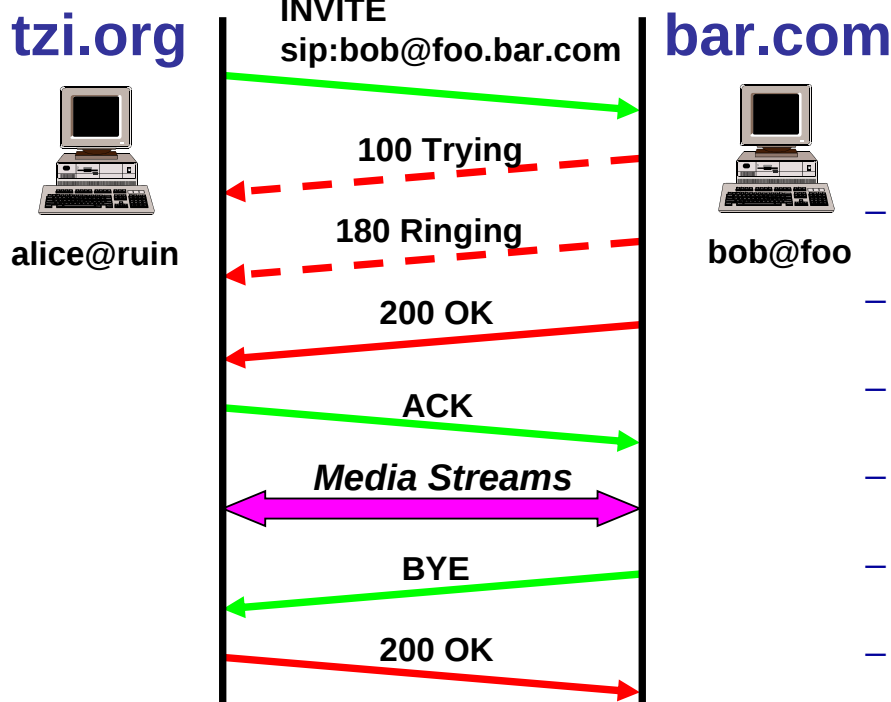
v=0
o=jdoe 28342 98543601 IN IP4 134.102.20.22
s=SIP call
t=0 0
c=IN IP4 134.102.20.38
m=audio 61002 RTP/AVP 0 4

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:

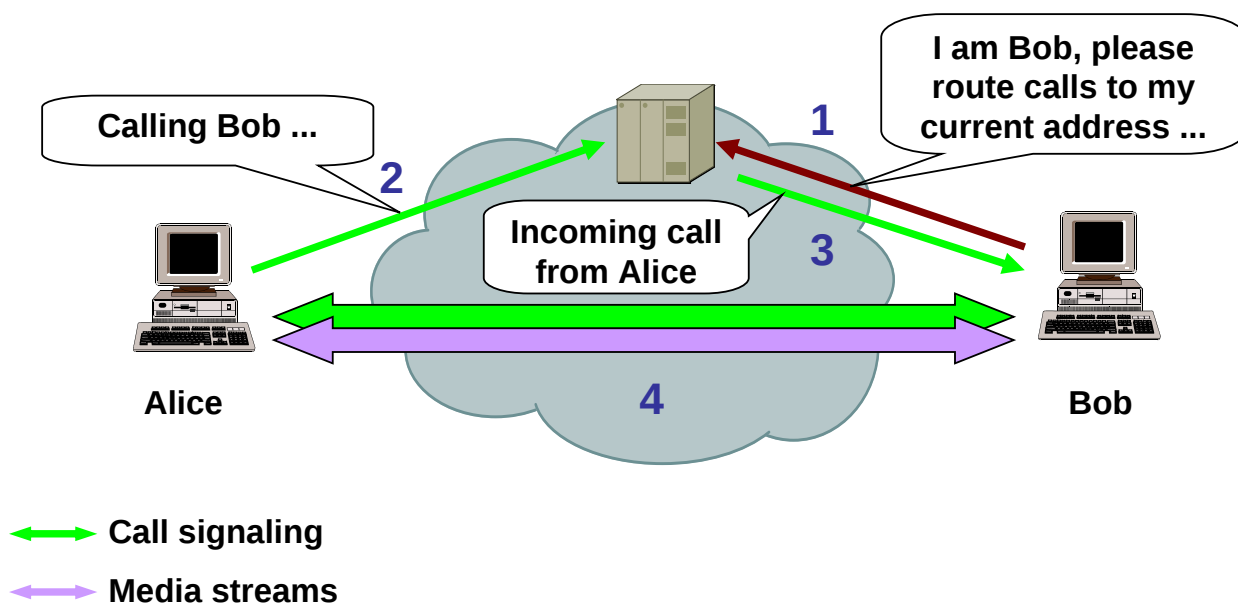
sip:user@domain

- 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

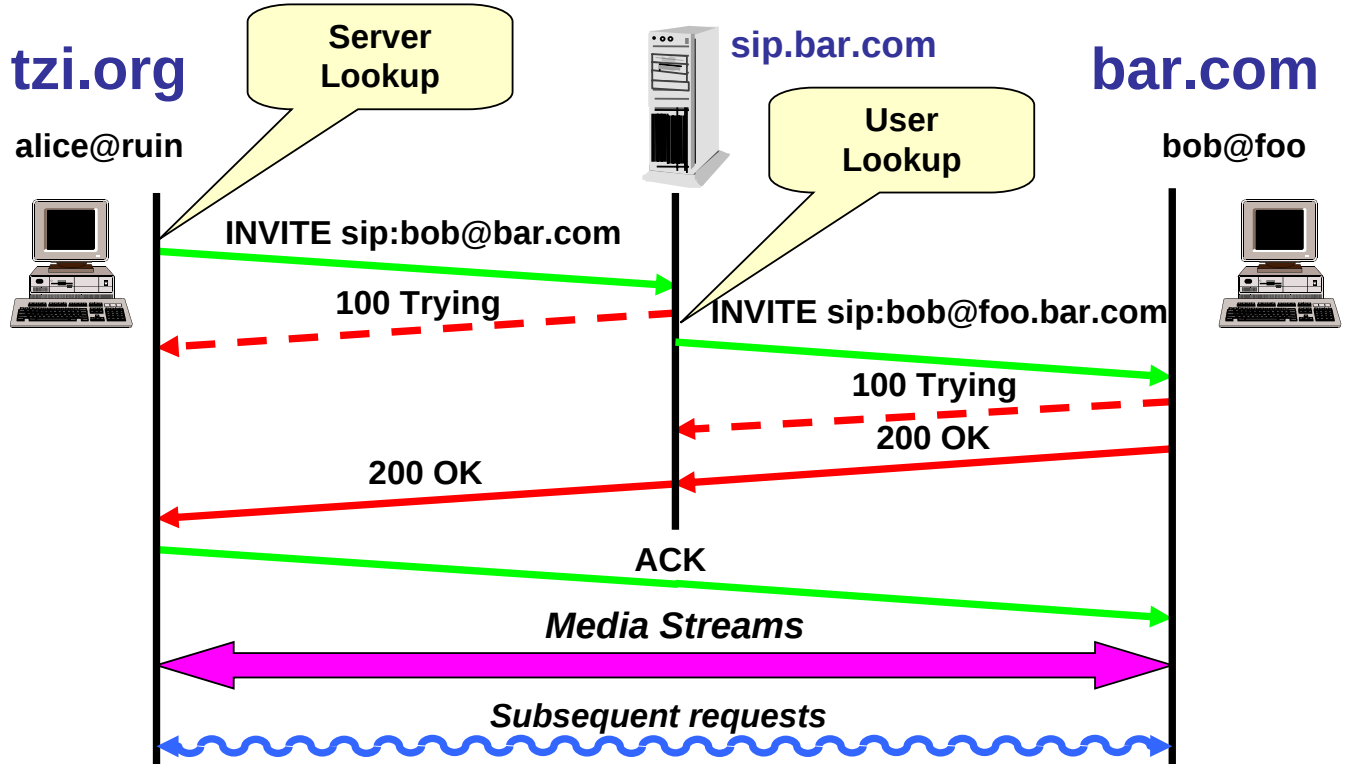
Finding the Next Hop

- ▶ **UAC may use a (manually) configured outbound proxy**
 - Outbound proxy may also have been 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*)

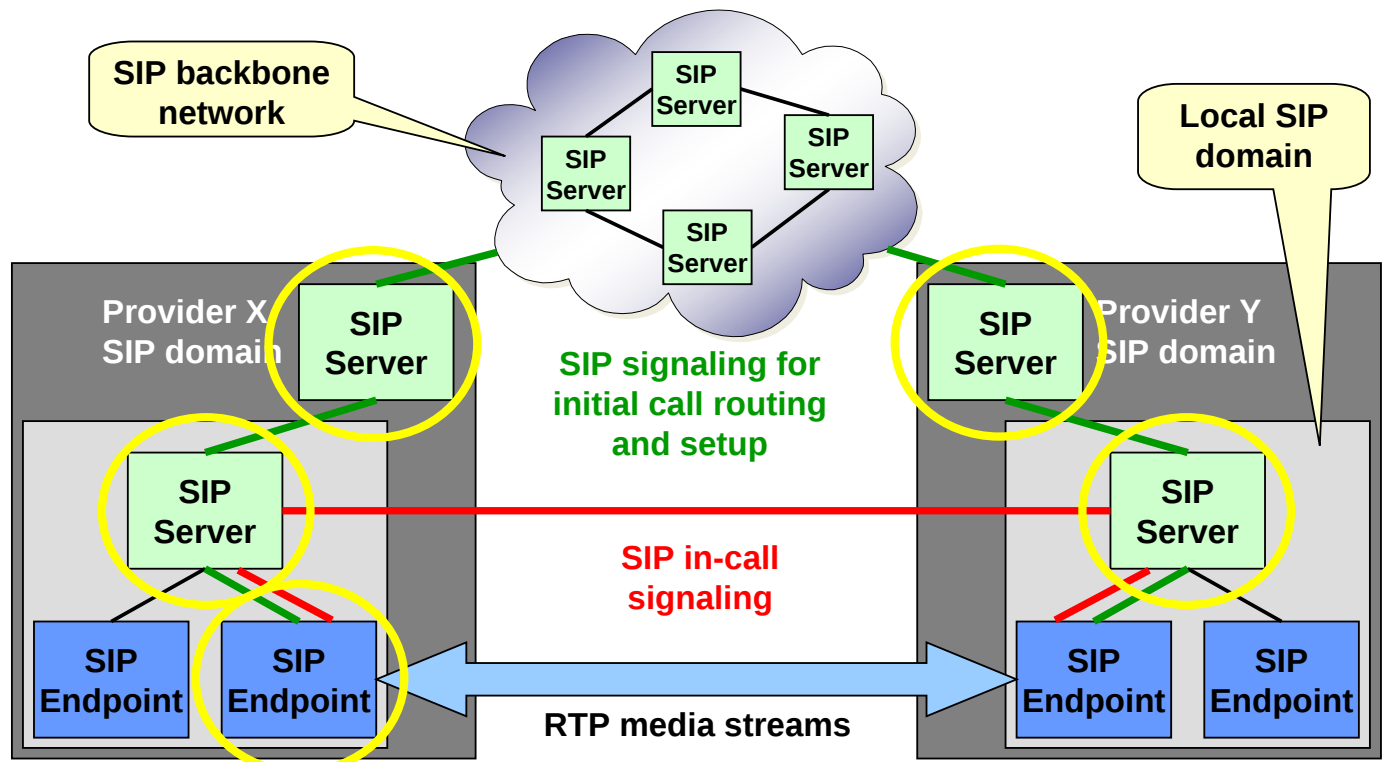
Simple Scenario of a SIP Call



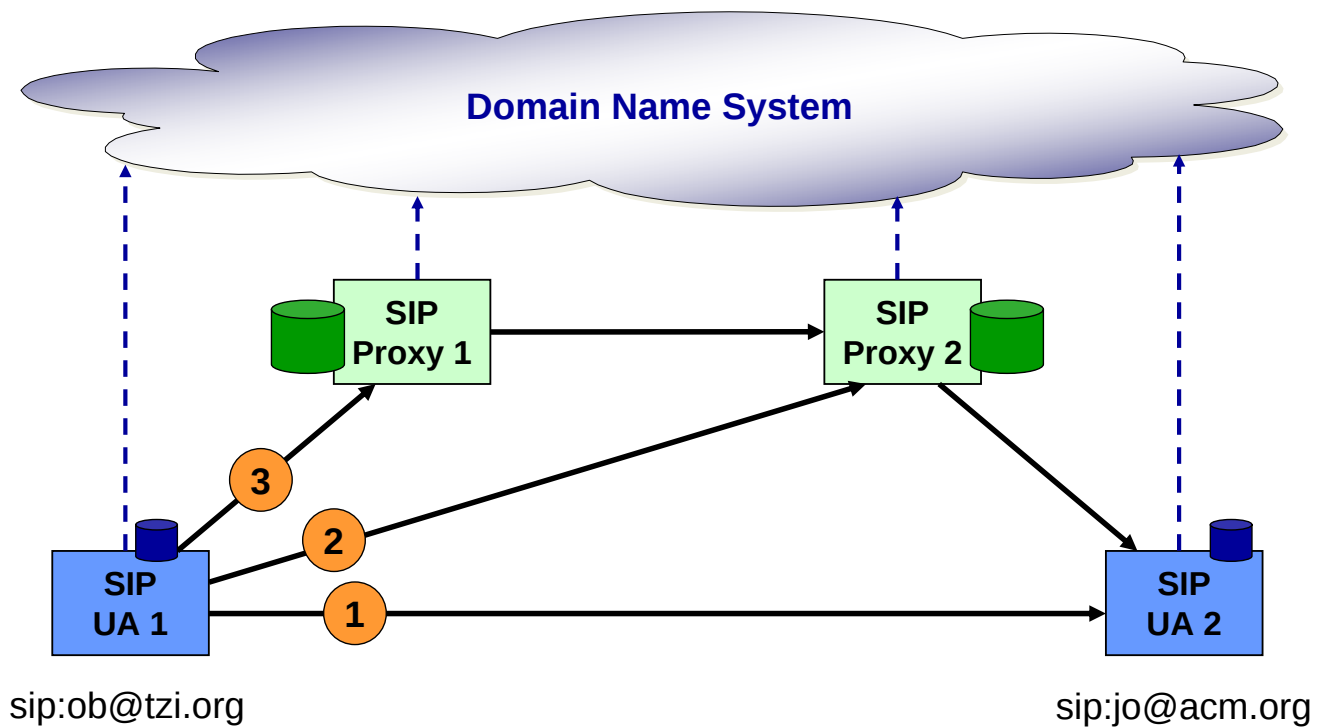
Proxied Call



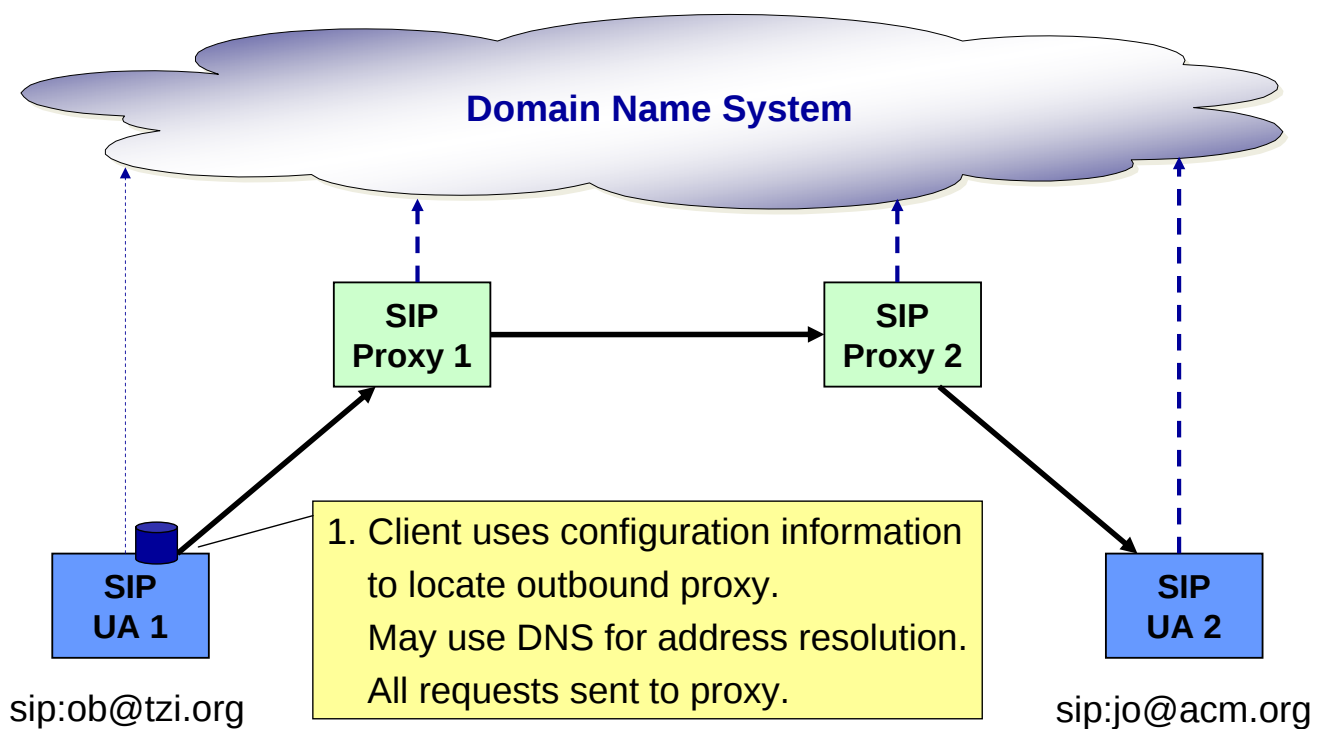
Global SIP Architecture



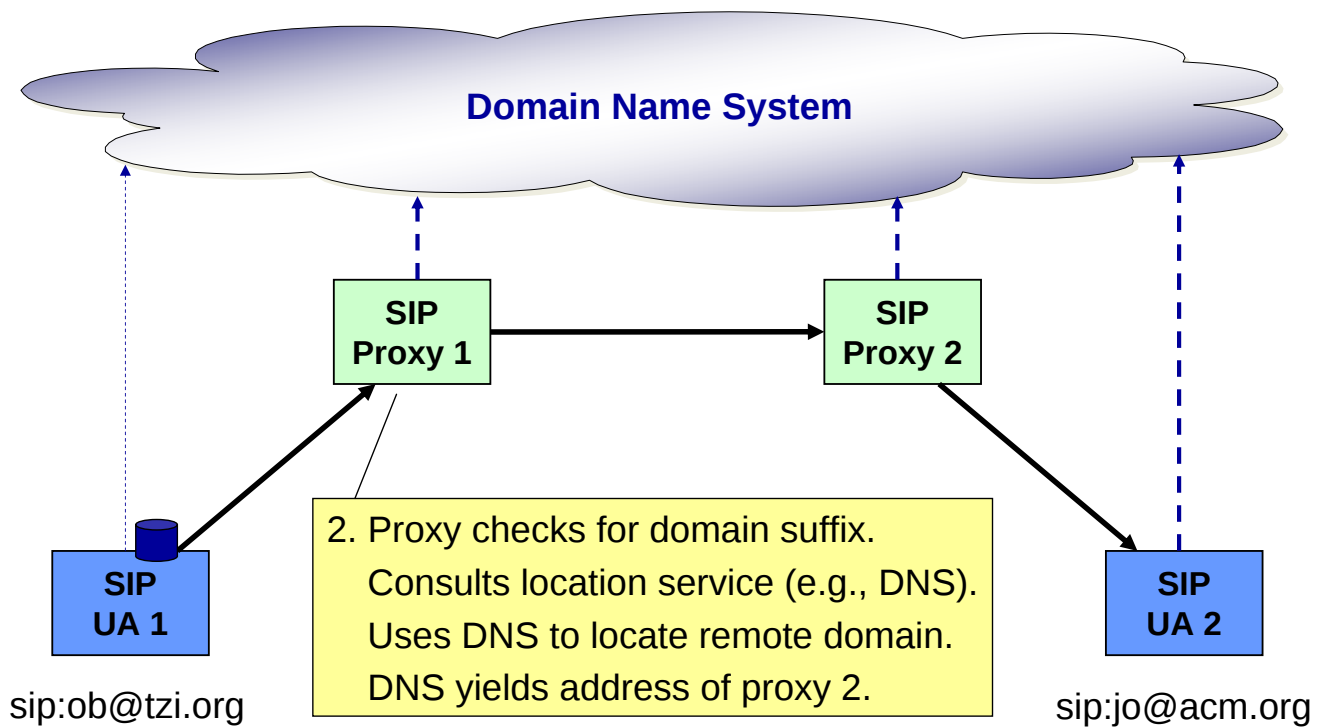
SIP Address Resolution Steps



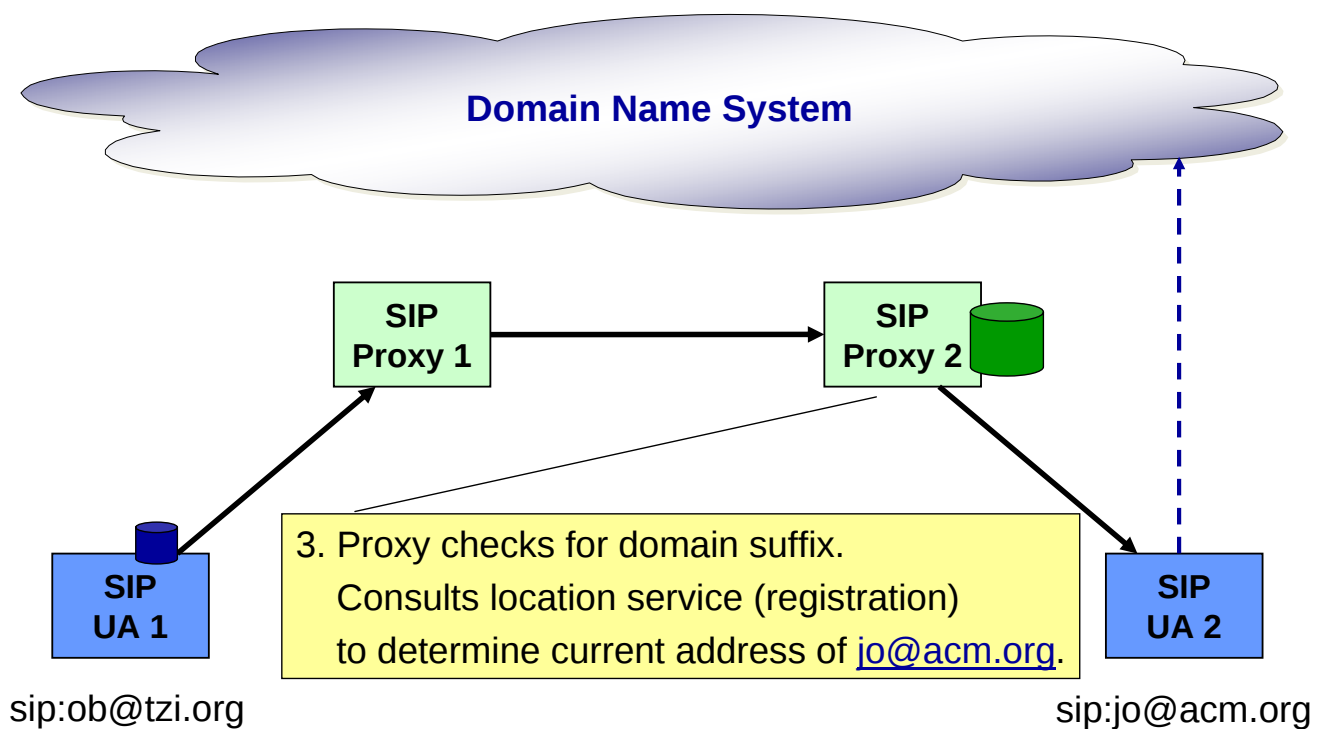
SIP Address Resolution Steps



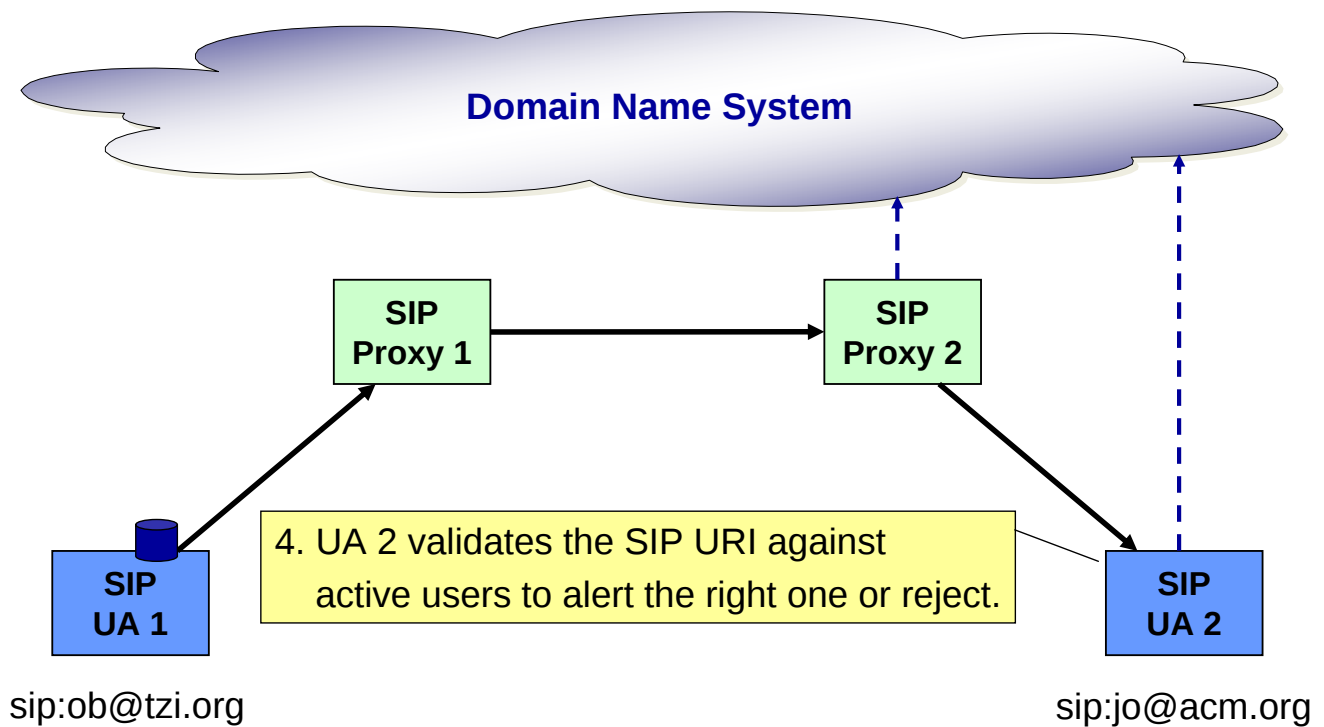
SIP Address Resolution Steps



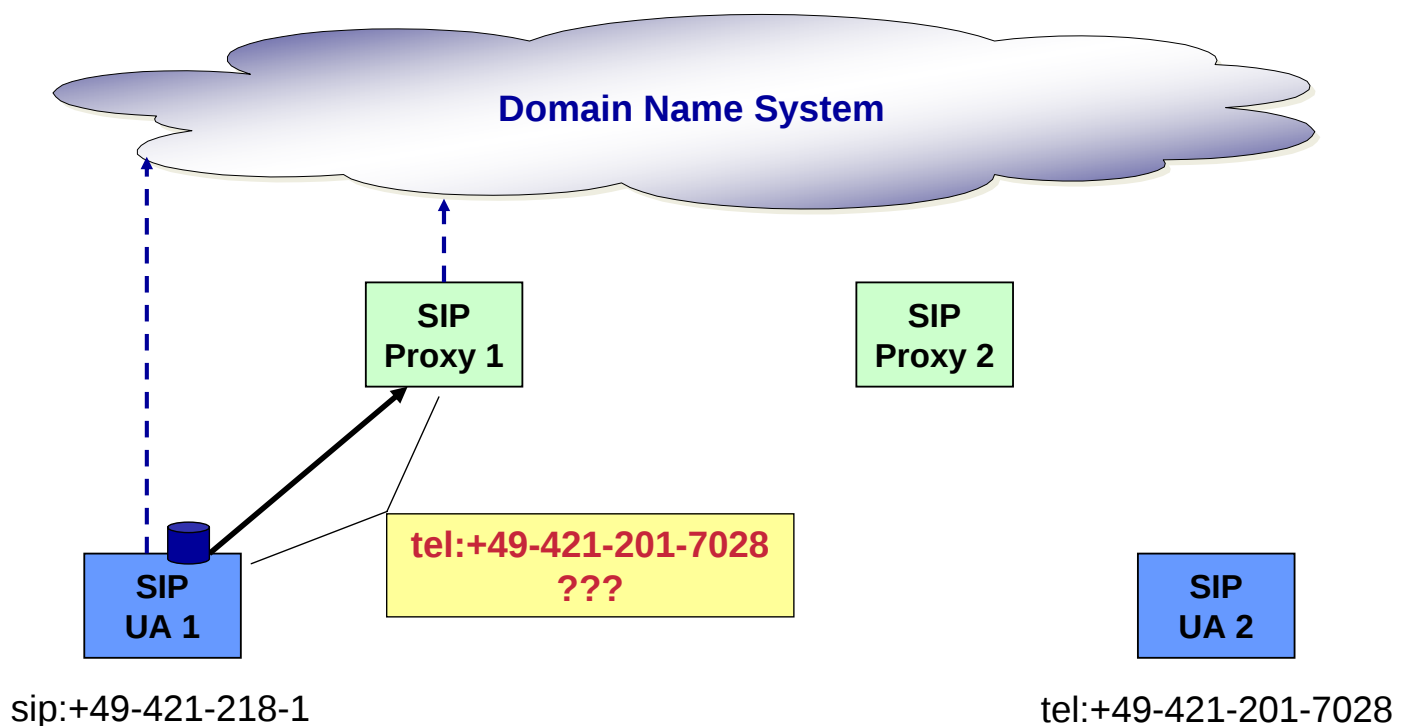
SIP Address Resolution Steps



SIP Address Resolution Steps



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 been learned upon registration

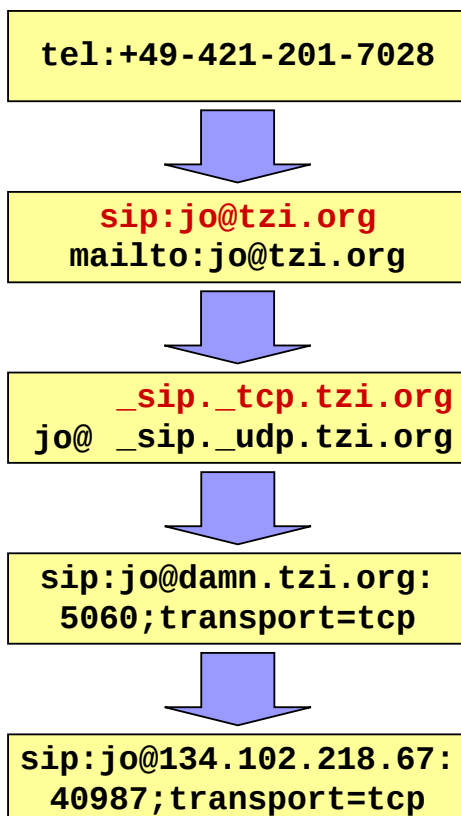
► If request URI contains IP address and port, message can be sent directly

► 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
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► **Query A or AAAA records for IP address**

ENUM Process

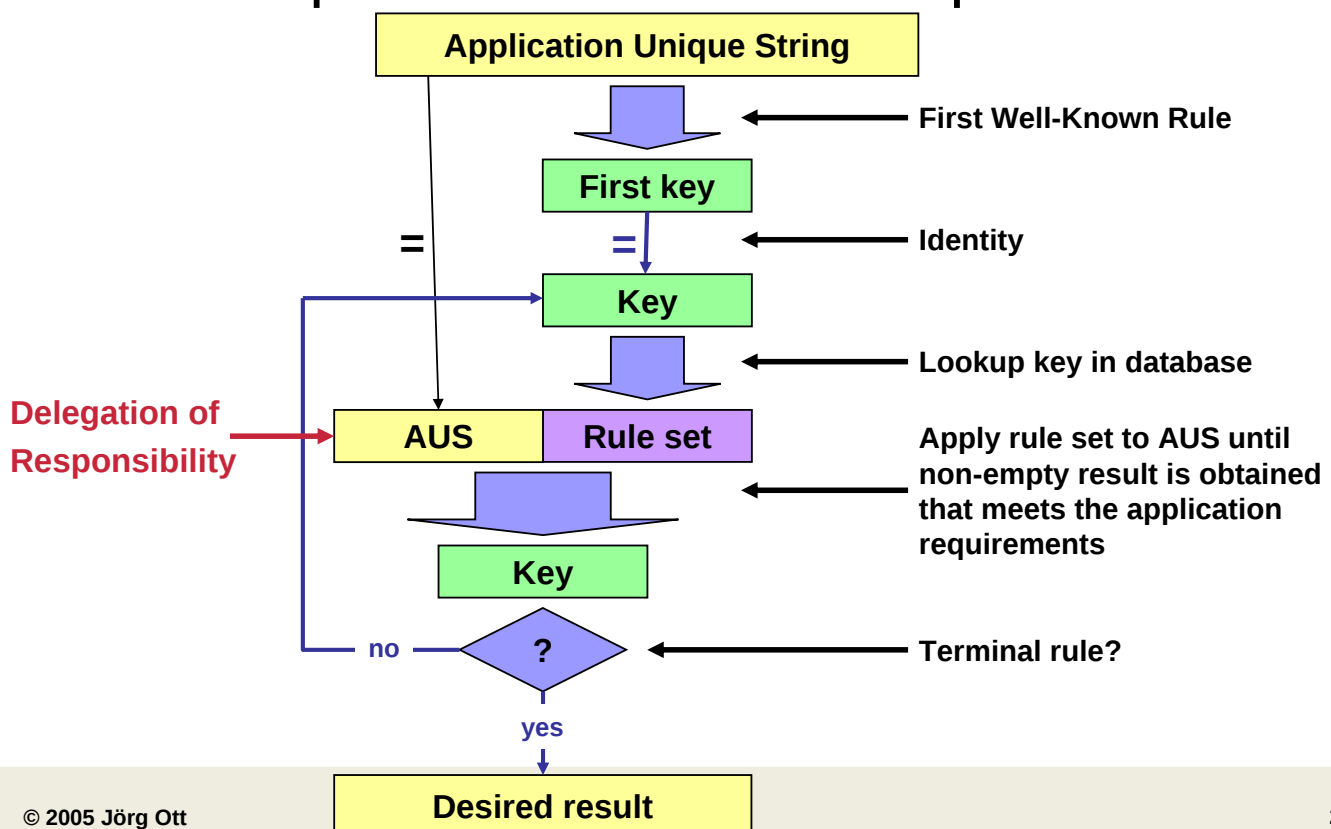


- ENUM service lookup
- Uses DNS NAPTR Resource Records
- NAPTR RR lookup to select preferred SIP services
- Based upon transport protocols and TLS
- SRV RR lookup for load balancing
- May or may not yield IP address
- A or AAAA lookup to determine IP address(es) associated with name

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 (regex-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

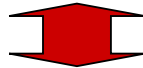


Step 1: NAPTR Resource Records

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

Order	Preference	Flags	Services	regexp	Replacement
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- ▶ 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
- ▶ Replacement: replacement for DNS name (= key)



Mutually exclusive

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
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- ▶ 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



Step 2: SIP NAPTR for Transport Selection

Type	Order	Preference	Flags	Services	regexp	Replacement
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- ▶ 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

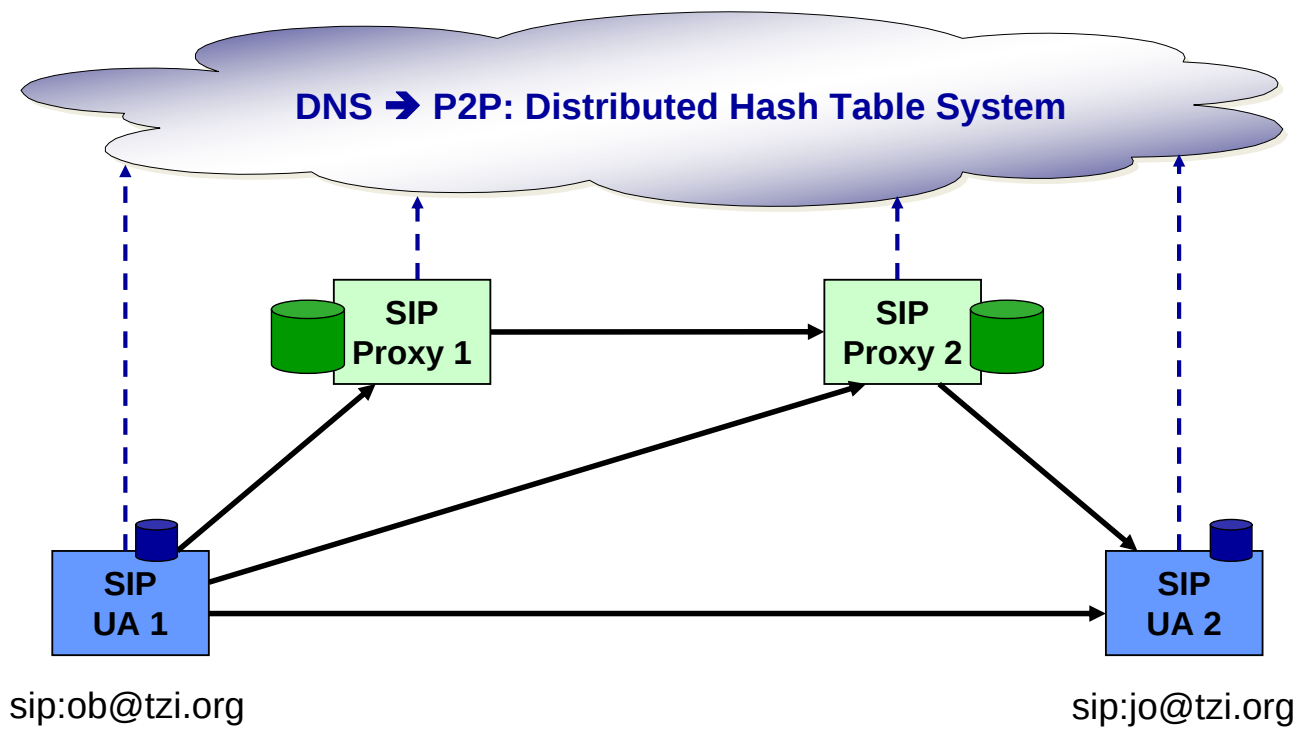
- ▶ Finally: lookup of A records for rasen.tzi.org
- ▶ Then send SIP message to 134.102.218.67



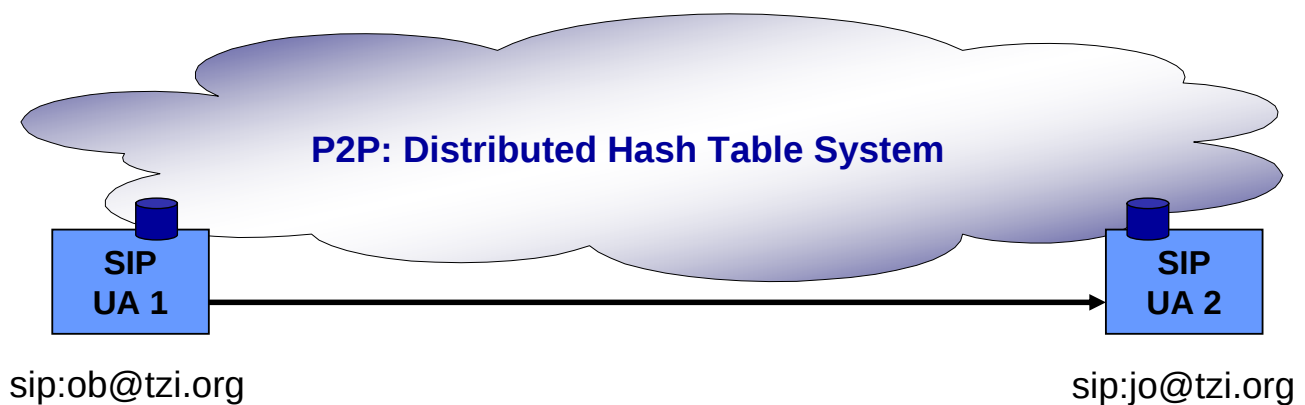
Alternative Address Resolution Schemes

- ▶ 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
- ▶ ...

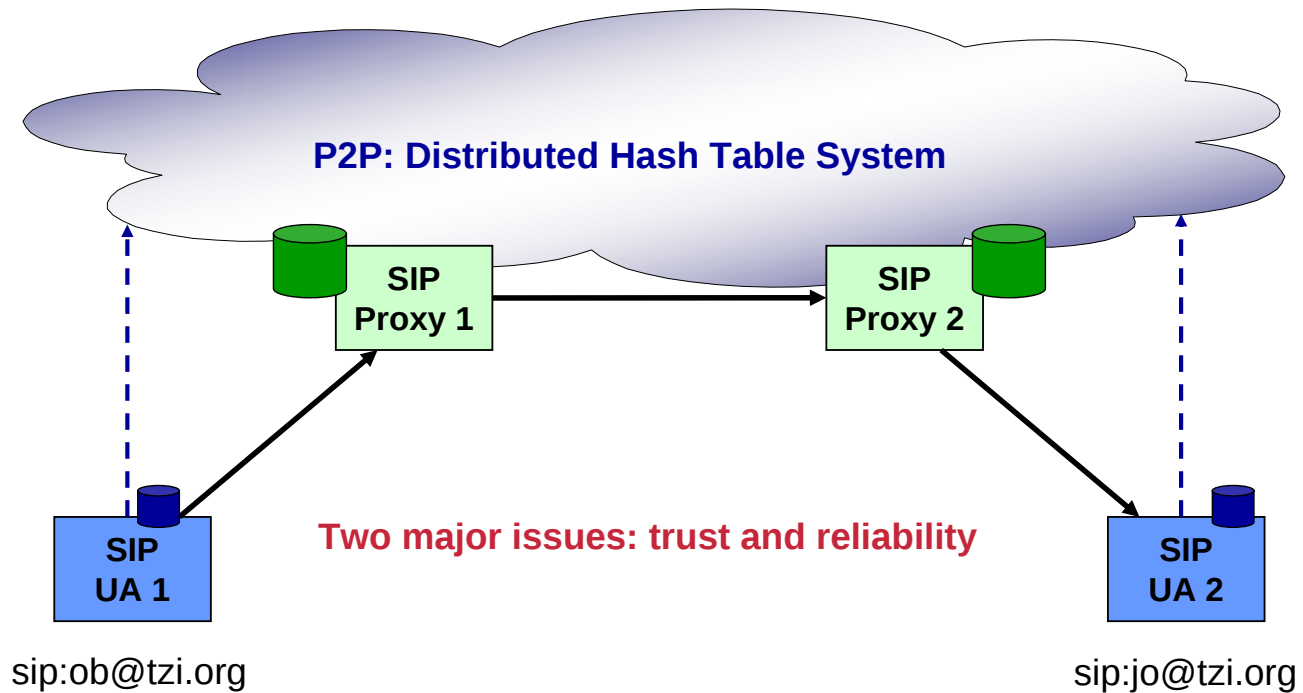
Peer-to-Peer SIP



Peer-to-Peer SIP: UAs as P2P Nodes



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