SIP and Mobility: IP Multimedia Subsystem in 3G Release 5

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Overview

- IETF Conferencing Architecture
- SIP Architecture and Basic Call Flows
- SIP Building Blocks for Services

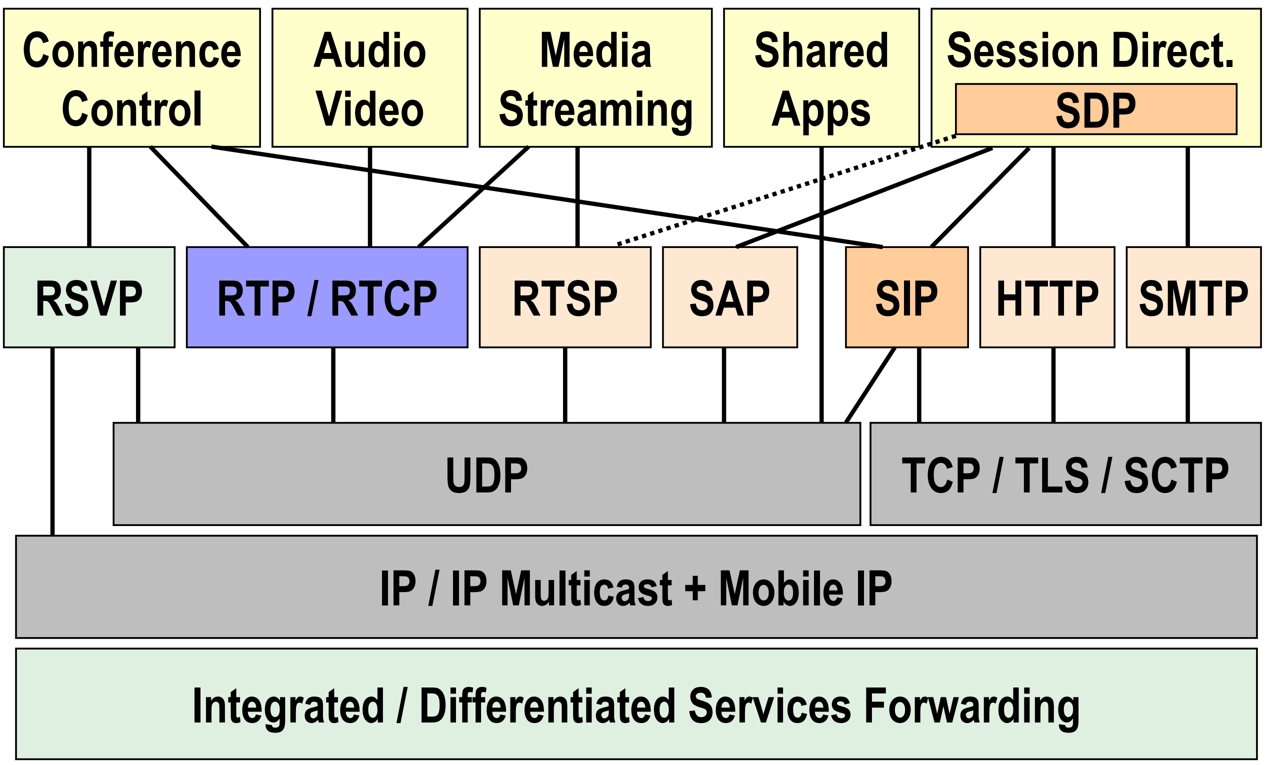
- 3G IP Multimedia Subsystem (IMS)
- Use of SIP in 3G
- 3G-Specific Extensions

- Conclusion

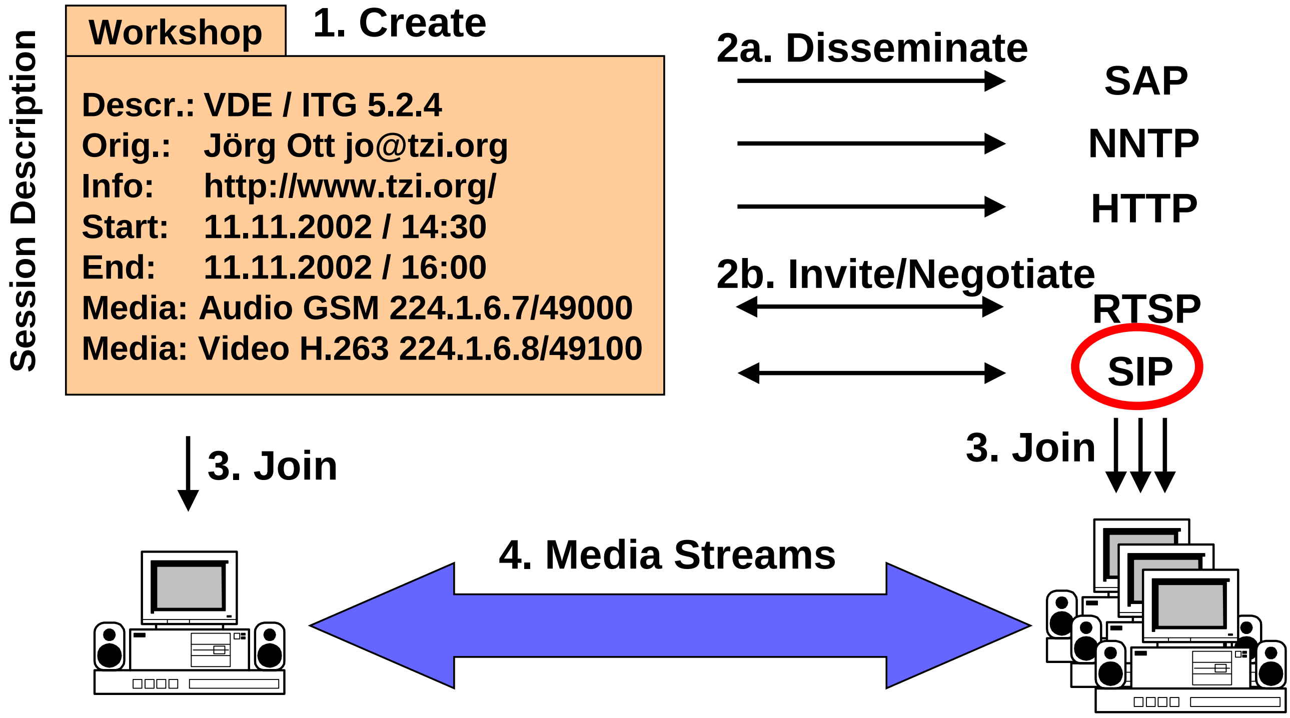
Mobility in (S)IP-enabled Environments

- **Ubiquitous communication environment providing (ideally seamless, uninterrupted) end-to-end services**
  - Independent of user location, device(s), access networks, …
- **User Mobility**
  - Users can roam and communicate from wherever they are
  - "Receive and place calls"
  - Users may use different devices in different places
- **Service Mobility**
  - Users can access the same service from different devices/locations
  - Users may move a communication relationship across devices
- **Device Mobility**
  - User devices may move across networks
  - retain and/or re-establish communication services

IETF Conferencing Architecture



IETF Conferencing Model



History of Mbone conference initiation

Session Invitation Protocol

(Handley/Schooler)

- Participant location
- Conference invitation
- Capability negotiation during setup

Simple Conference Invitation Protocol

(Schulzrinne)

- Participant location
- Conference invitation
- Capability negotiation during setup
- Changing conference parameters
- Terminate/leave conference

1996

Session Initiation Protocol

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 for presence and instant messaging

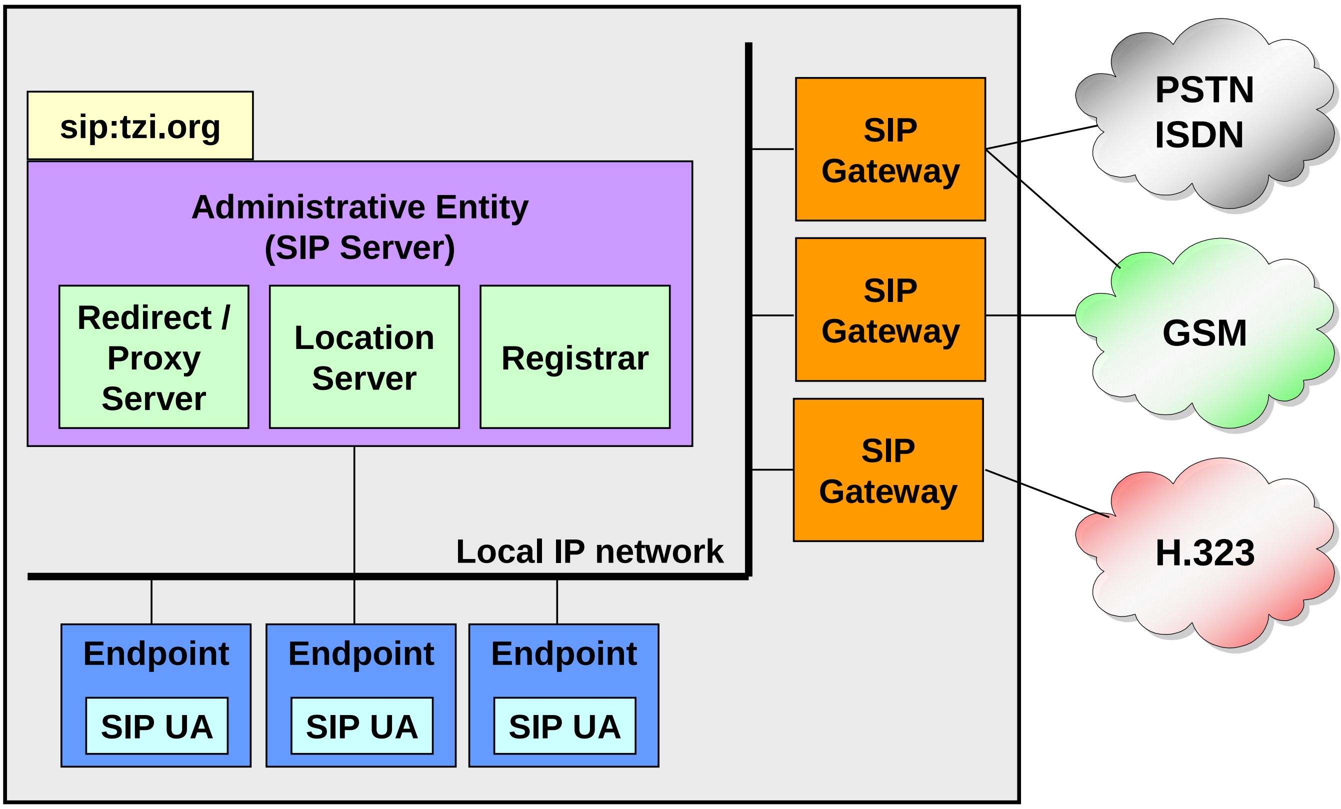
Terminology

- **User Agent Client (UAC):**
  - Endpoint, initiates SIP transactions
- **User Agent Server (UAS):**
  - Handles incoming SIP requests

User Agent

- **Redirect server:**
  - Retrieves addresses for callee and returns them to caller
- **Proxy (server):**
  - UAS/UAC that autonomously processes requests
    → forward incoming messages (probably modified)
- **Registrar:**
  - Stores explicitly registered user addresses
- **Location Server:**
  - Provides information about a target user's location
- **Back-to-Back User Agent (B2BUA)**
  - Keeps call state; more powerful intervention than proxy

Local SIP Architecture



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
```


SIP URI Addressing Scheme 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 or group; typically contains host name or IP address, port, transport protocol, ...
- **URIs may carry additional parameters**

´sip:´ [*user* [´:´ *passwd*] ´@´] *host* [´:´ *port*] *params* [´?´ *headers*]

params ::= (´;´ *name* [´=´ *value*])*
headers ::= *field* ´=´ *value*? [´&´ *headers*]

- **URIs may also identify services**

SIP URI Addressing Examples

```
sip:tzi.org
sip:192.168.42.1
```

Registration domain or IP address

```
sip:john@example.com
sip:john@example.com:5060
sip:foo@example.com:12780
```

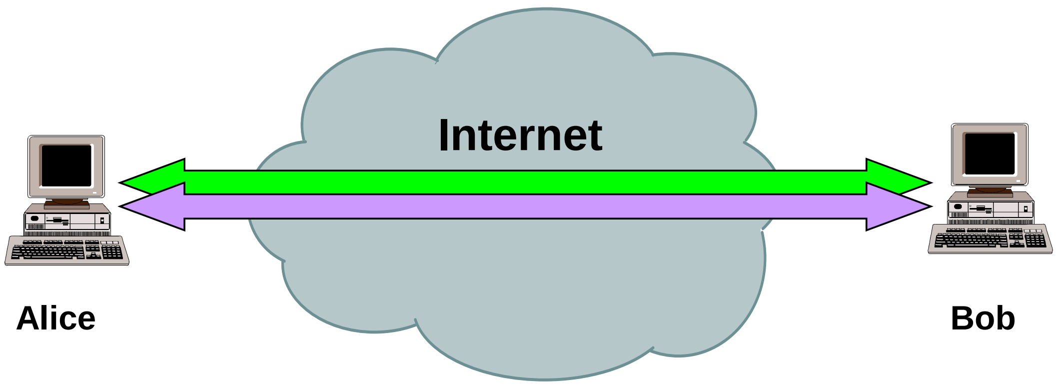
Userinfo + domain/host optional port number

```
sip:voicemail@service.com
sip:conf-1234@confserv.com
sip:user34@anonymizer.org
```

Service identifier; semantics opaque to the user

Use URI scheme ‘sips’ to request secure communications.

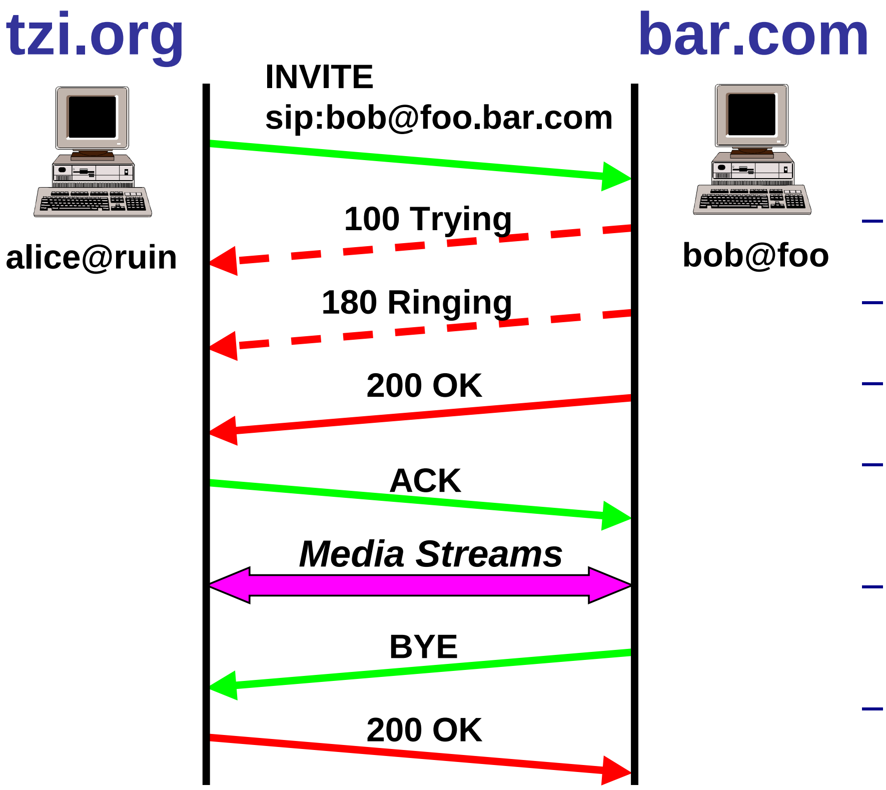
Application Scenario 1: Direct Call UA–UA



Call signaling

Media streams

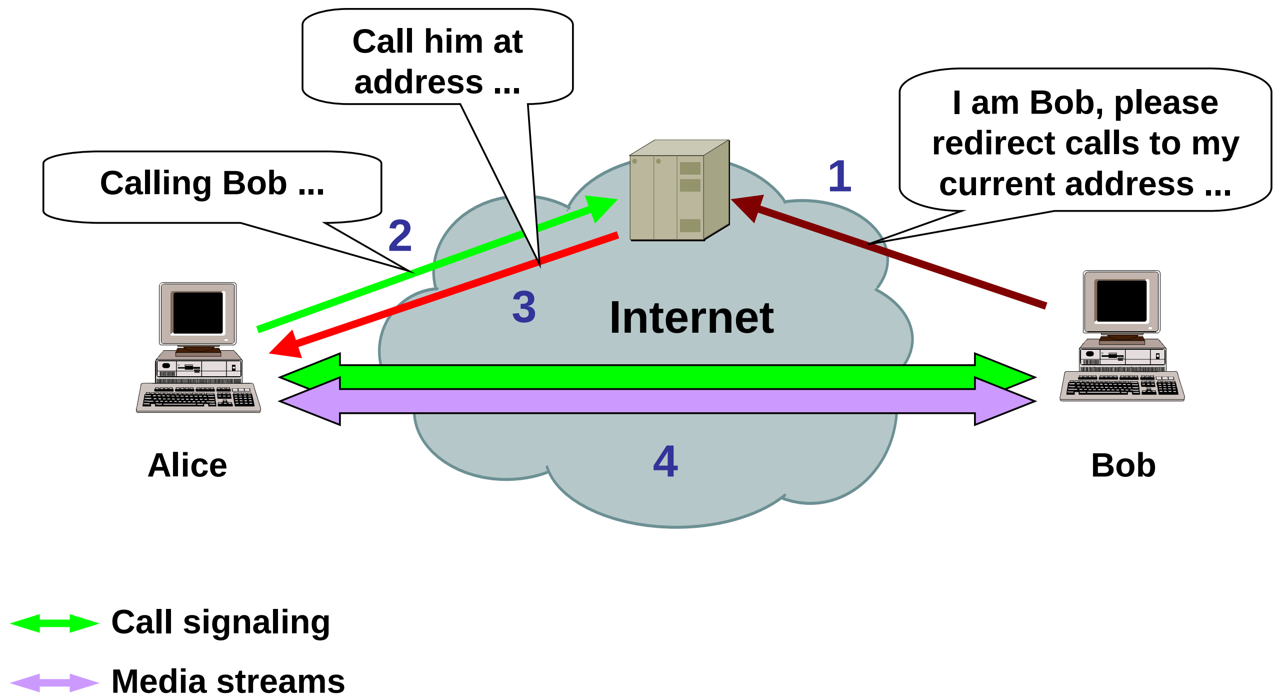
Direct Call



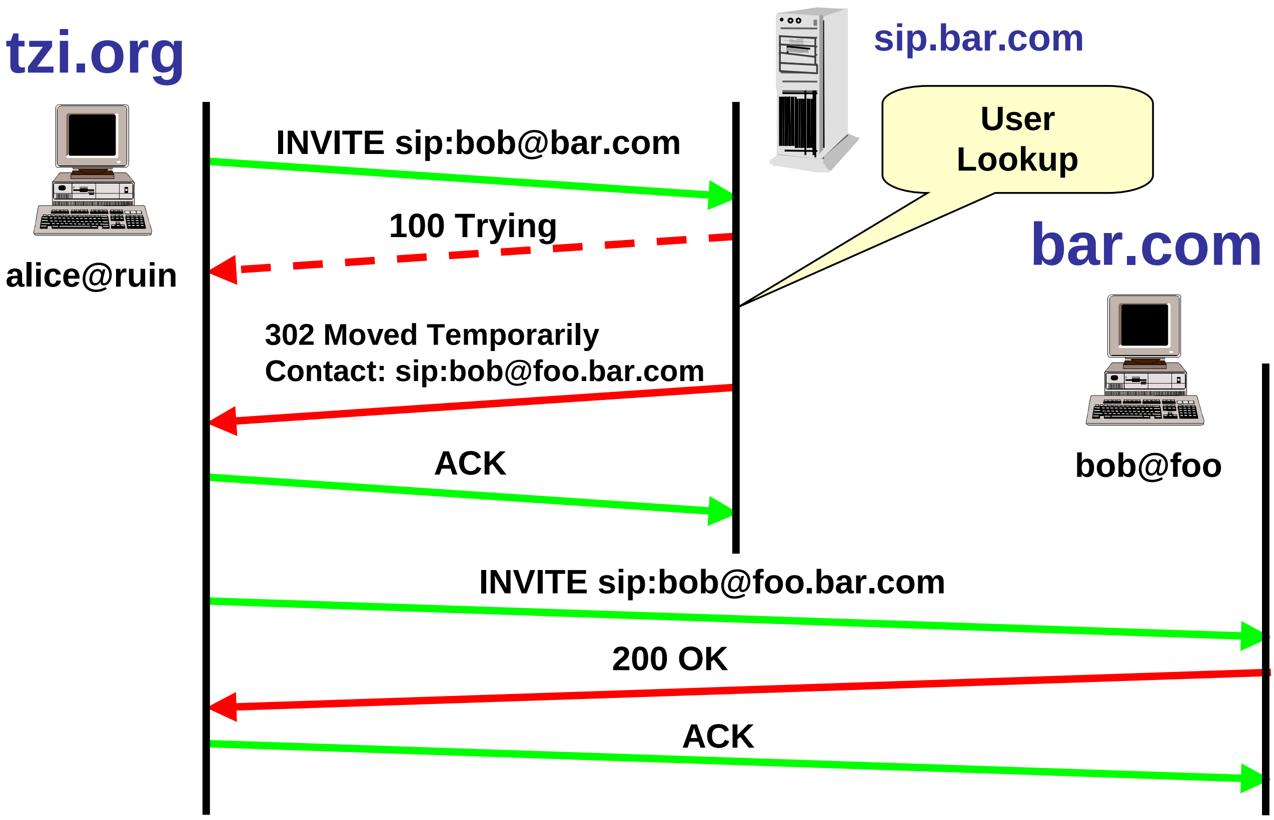
Note:
Three-way handshake is performed only for INVITE requests.

- 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

Application Scenario 2: Redirected Call



Redirected Call



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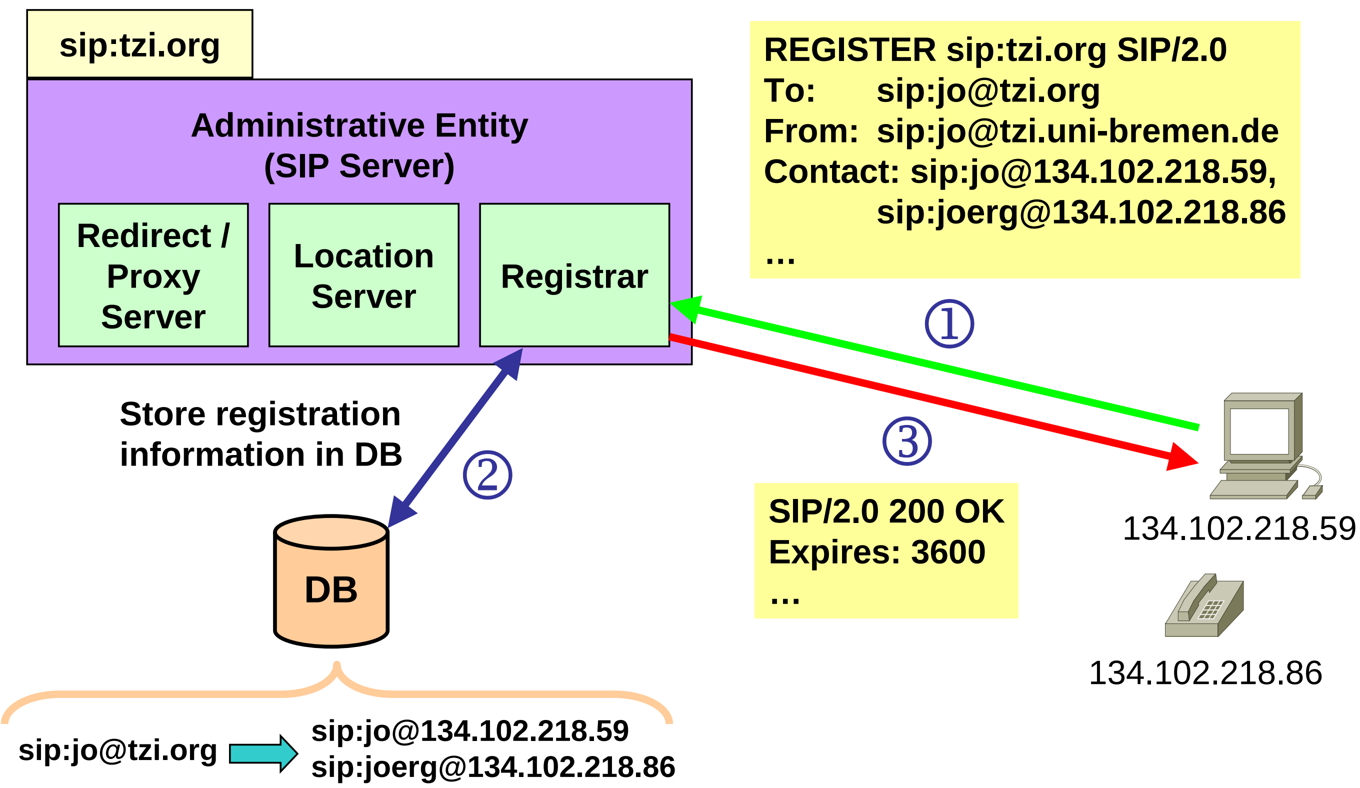
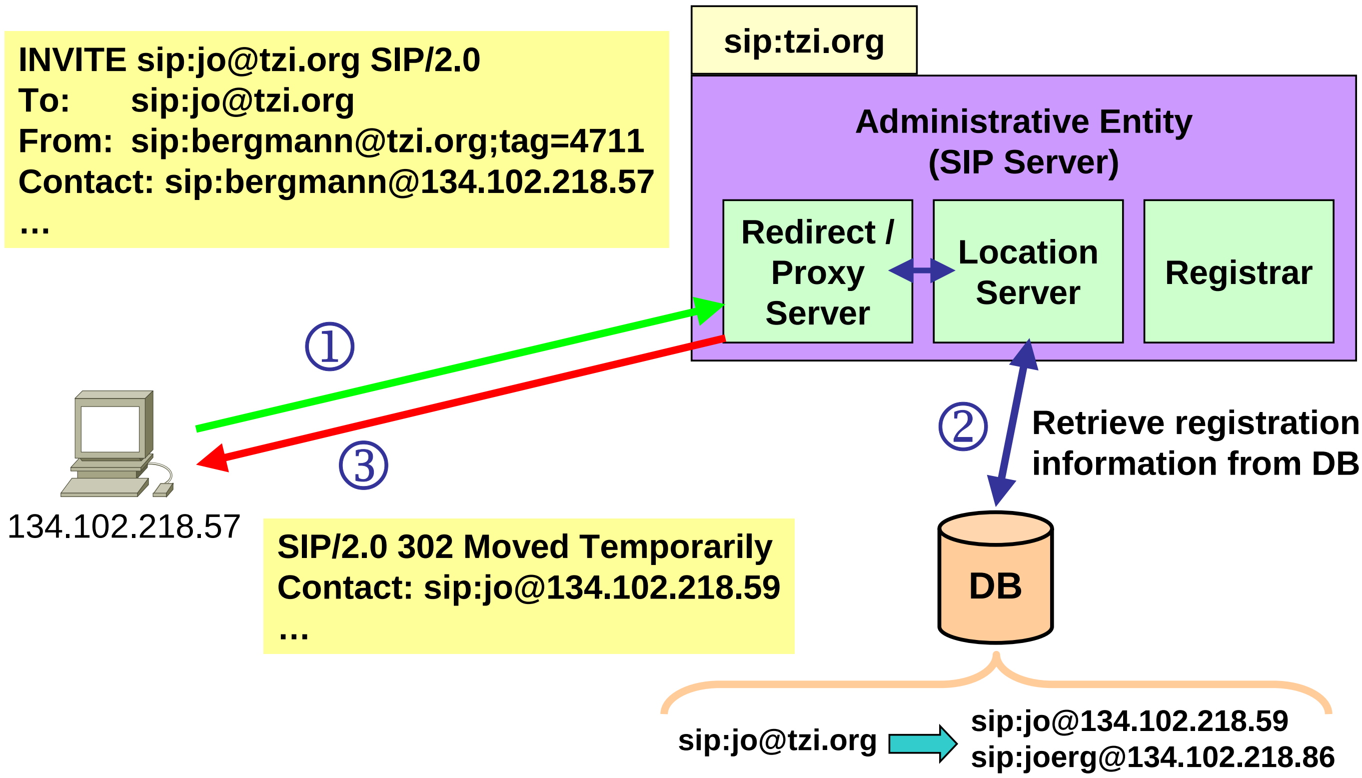
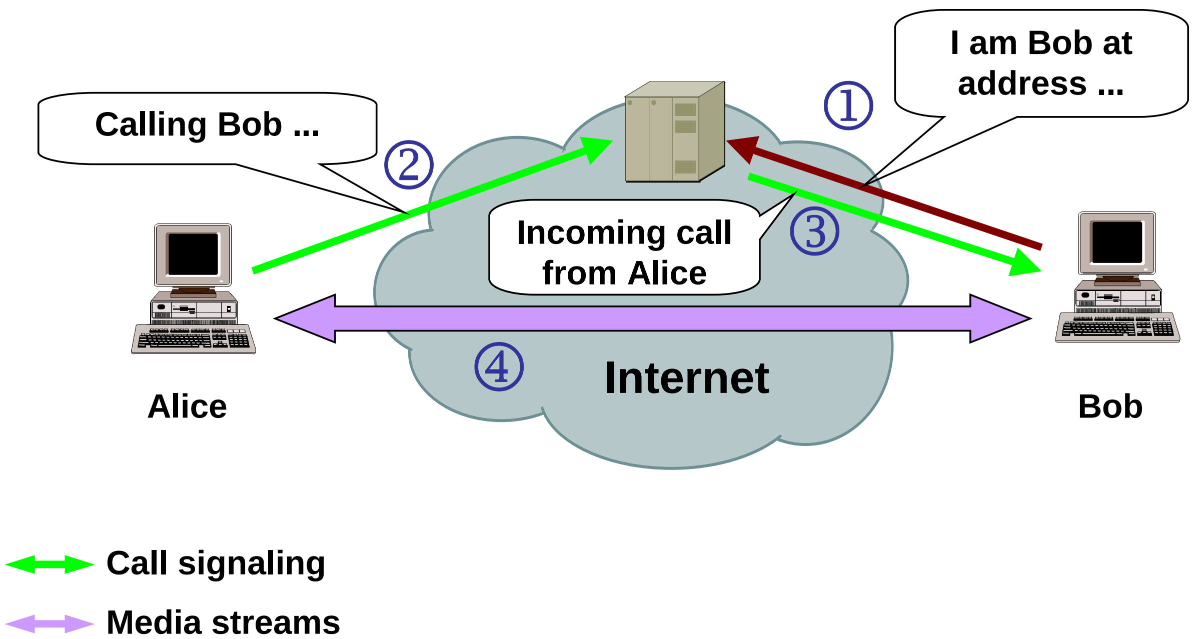
User Registration



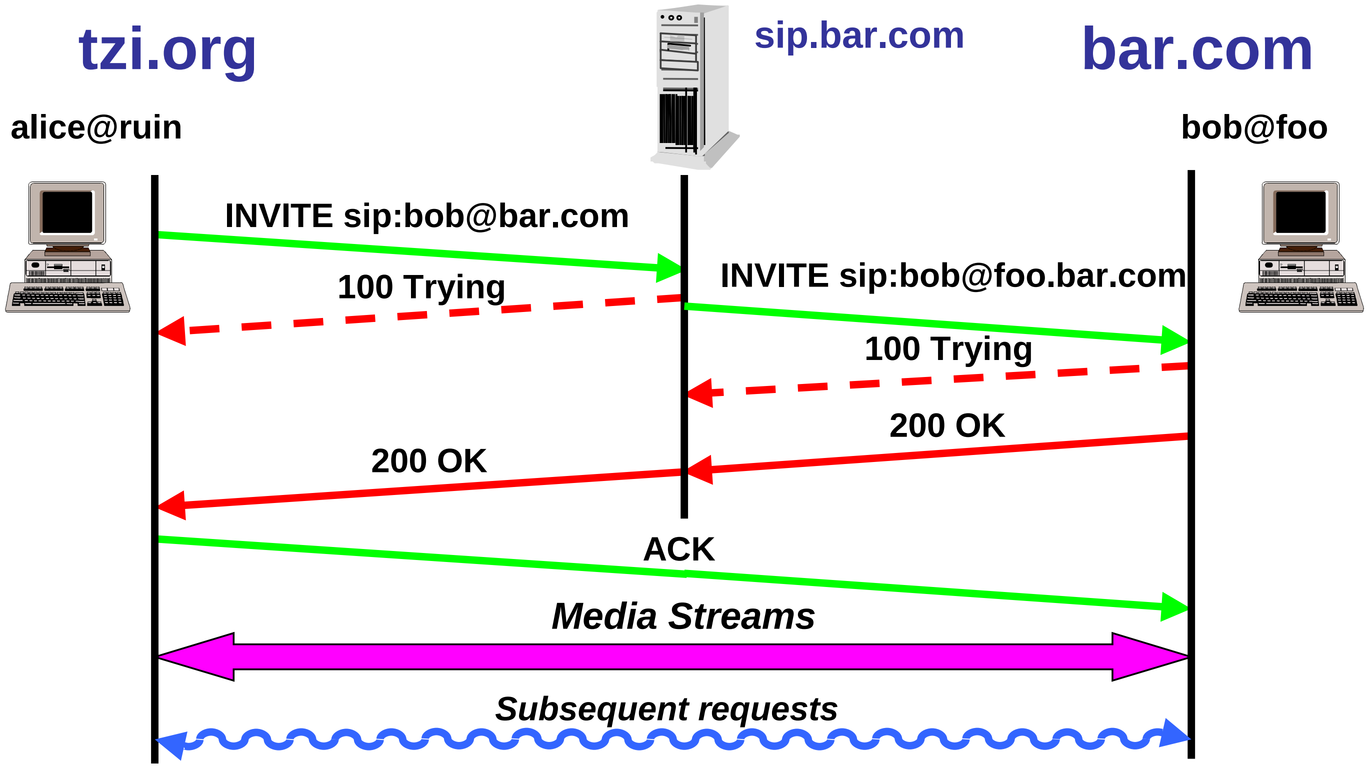
User Location



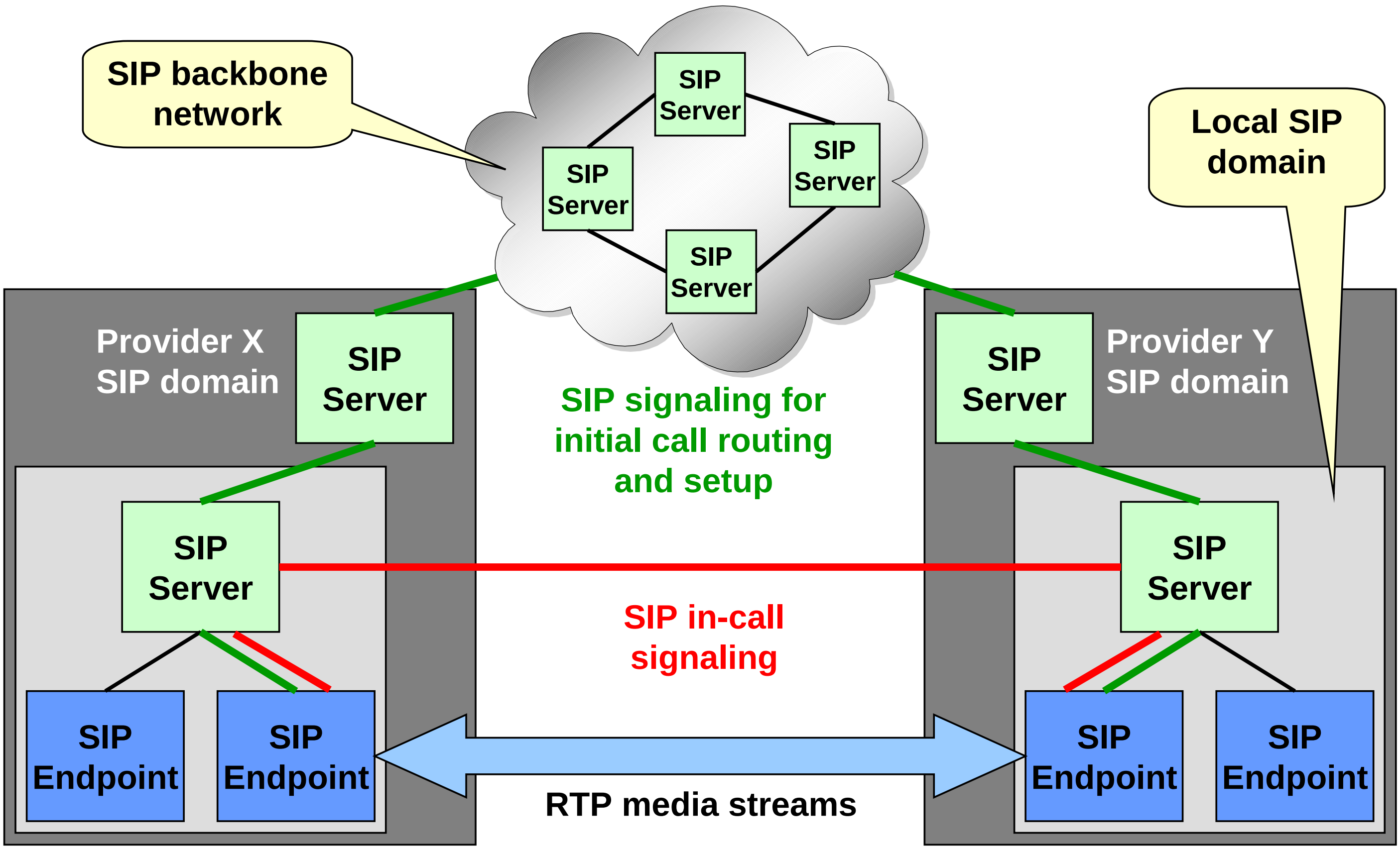
Application Scenario 3: Proxied Call



Proxied Call



Inter-Domain (“Real World”) SIP Architecture



SIP Proxy Functionality

- **Stateless vs. stateful**
  - Stateless: efficient and scalable call routing
  - Stateful: service provision, firewall control, ...

Some roles for proxies

- **Outbound proxy**
  - Perform address resolution and call routing for endpoints
  - Pre-configured for endpoint (manually, DHCP, ...)
- **Backbone proxy**
  - Essentially call routing functionality
- **Access proxy**
  - User authentication and authorization, accounting
  - Hide network internals (topology, devices, users, etc.)
- **Local IP telephony server (IP PBX)**
- **Service creation in general...**

Service Creation in SIP

- **Points of Service in SIP**
  - Endpoints!
  - SIP Servers (proxies, redirect, location servers, …)
  - SIP Application Servers
  - Back-to-Back User Agents
- **May be located anywhere**
  - in the “network”, controlled by the service provider
  - “outside the network”, controlled by third party providers
  - at the “user premises”, controlled by the user
  - and any combination of the above!
- **May be combined in a flexible manner**
- **Administrative services and user services**
  - Focus on server-based services

SIP Administrative Services

Examples

- User Authentication and Authorization
  - Proxy communicates with back-end servers (e.g. via RADIUS)
  - Proxy retrieves certificates from LDAP servers (PKI)
  - Proxy checks user's permissions + credit + access / call rules
- Accounting
  - Call-stateful proxies log call duration, produce Call Detail Records
- User Location
  - Basic input from registration database (registrar)
  - Use of presence information as additional cues
  - Consideration of caller preferences
  - Call handling per callee rules (e.g. CPL)
- Call Services
  - Malicious call tracing

SIP User Services

Examples

- Incoming call handling
  - Based upon calling user, time of day, day of week, …
  - Based upon current presence status
- Answering Machine
  - Redirect call to media server
- Call services
  - Call redirection, call deflection, ...
  - Caller identity
  - Anonymous calls
- Conferencing
  - Scheduled and ad-hoc conferencing

➔ May be implemented in proxies or application servers
  - Using SIP naming and routing logic to route requests

Proxy-based Service Creation

- **SIP proxies have limited capabilities**
  - They only forward (modified) messages
  - They cannot terminate calls
  - They MUST NOT initiate requests on their own

  → Impossible to enforce call termination via a proxy
  → Impossible to modify call state and involved parties

- **Concept to a Back-to-Back User Agent (B2BUA)**
  - Terminates and originates calls (instead of proxying)
  - Relates two or more dialogues at the "application layer"

- **Calls need to be routed to / through proxy or B2BUA**
  - User-initiated to obtain a certain service
  - Provider-enforced to retain a certain level of control

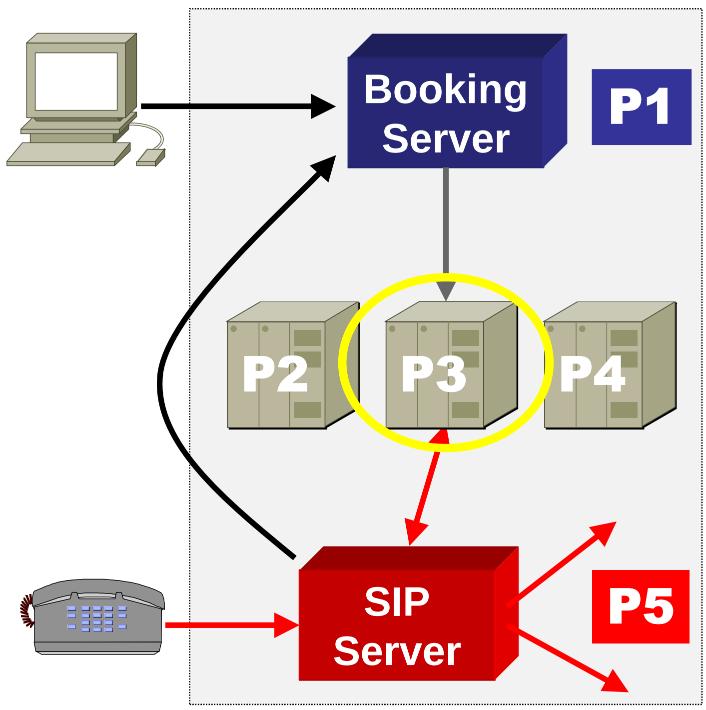
- **Outbound proxies, access proxies, "home" proxies, …**

B2BUA Service Creation Example: Conferencing

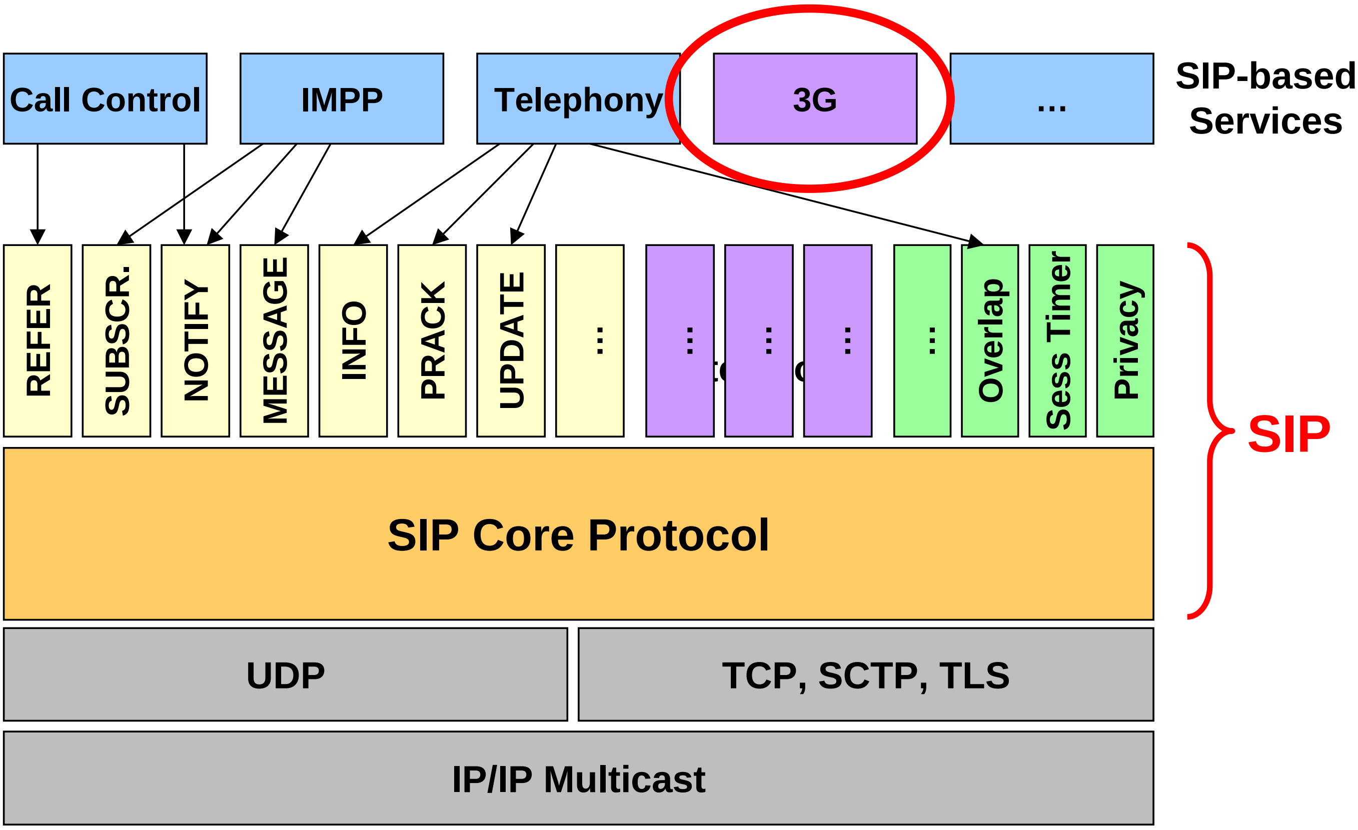
- **Provider 1 offers highly available conference booking services**
  - contracts with 2 – 5

- **Providers 2 – 4 offer conference bridges with different capabilities**

- **Provider 5 offers IP Telephony Services**
  - gets value add from P1
  - but also contracts others



SIP Features and Service Creation Model



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SIP and 3G

SIP Support for Mobility

Personal Mobility
- SIP URIs
- SIP registration and authentication
- SIP call routing
- SIP routing to services

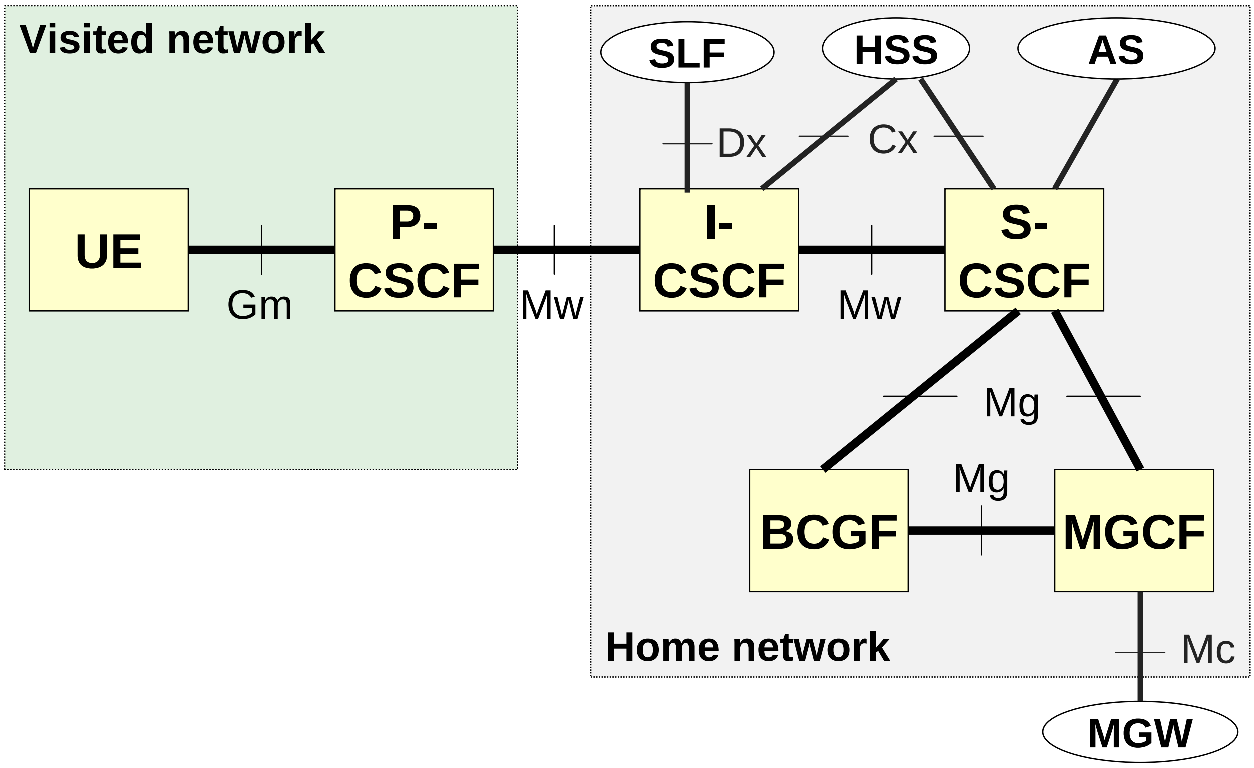
Terminal Mobility
- Ubiquitous IP connectivity (+ roaming)
- Mobile IP (orthogonal to SIP)
- Dynamic SIP session re-routing
- 3G link layer roaming mechanisms

Service Mobility
- Dynamic SIP session re-routing
- 3G Hand-off procedures

SIP and 3GPP

- 3G networks offer two modes of operation:
  - circuits and packets
- Multimedia functionality based on packets
  - IP Multimedia Subsystem (IMS) in the Core Network (CN)
  - (exception: H.324 used for video telephony)
- IMS uses SIP for signaling
  - one piece out of a number of IETF protocols
- In the long run, all services shall converge to IP
- Release 5: SIP-based multimedia calls
- Release 6: Further SIP services to come (e.g. Presence, IM)

SIP-related Components in 3GPP



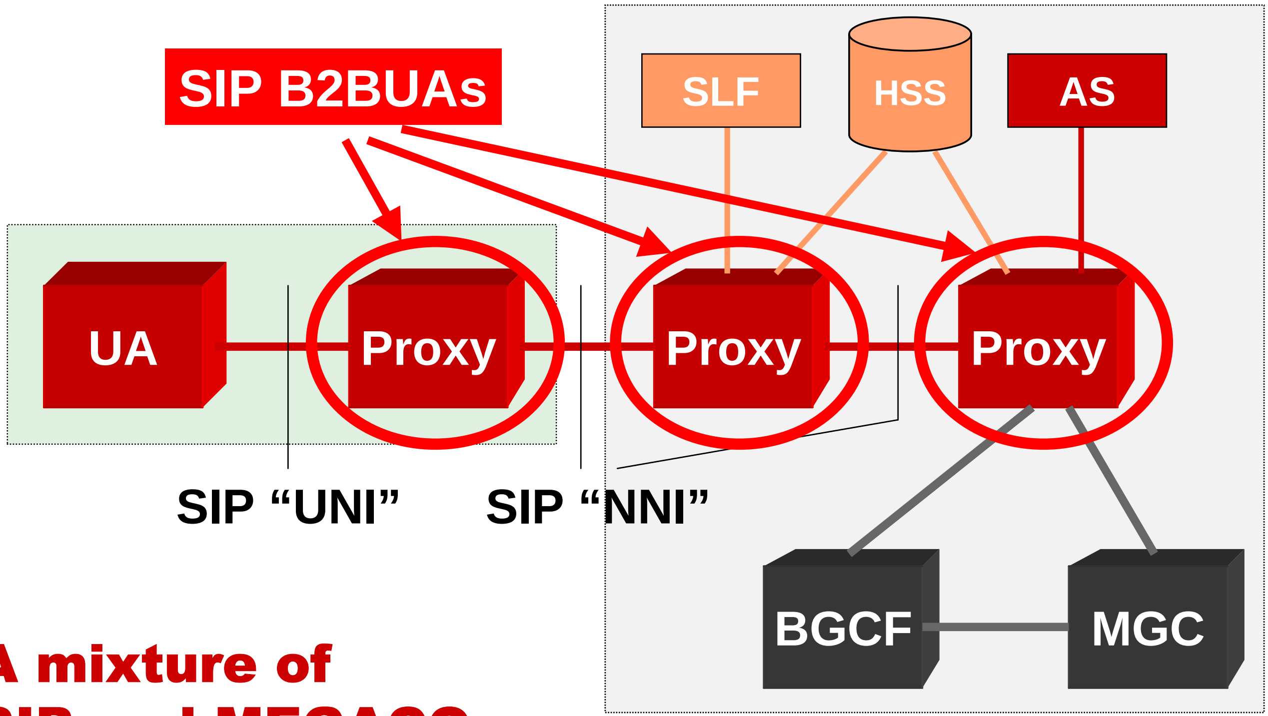
A Collection of Acronyms

- **UE** User Equipment

- **P-CSFC** Proxy Call Session Control Function
- **I-CSFC** Interrogating Call Session Control Function
- **S-CSFC** Serving Call Session Control Function
- **HSS** Home Subscriber Server
- **AS** Application Server
- **SLF** Subscription Locator Function

- **BGCF** Breakout Gateway Control Function
- **MGCF** Media Gateway Control Function
- **MGW** Media Gateway

SIP Components in 3GPP



A mixture of SIP and MEGACO…

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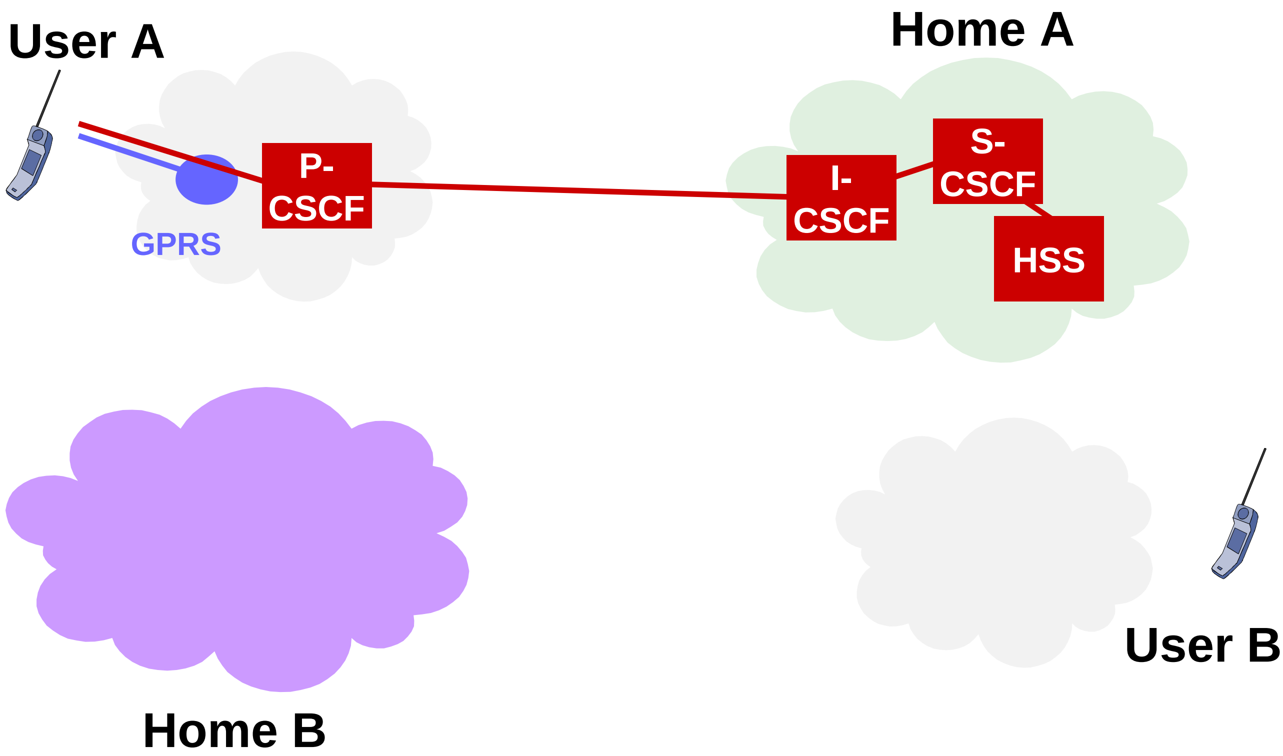
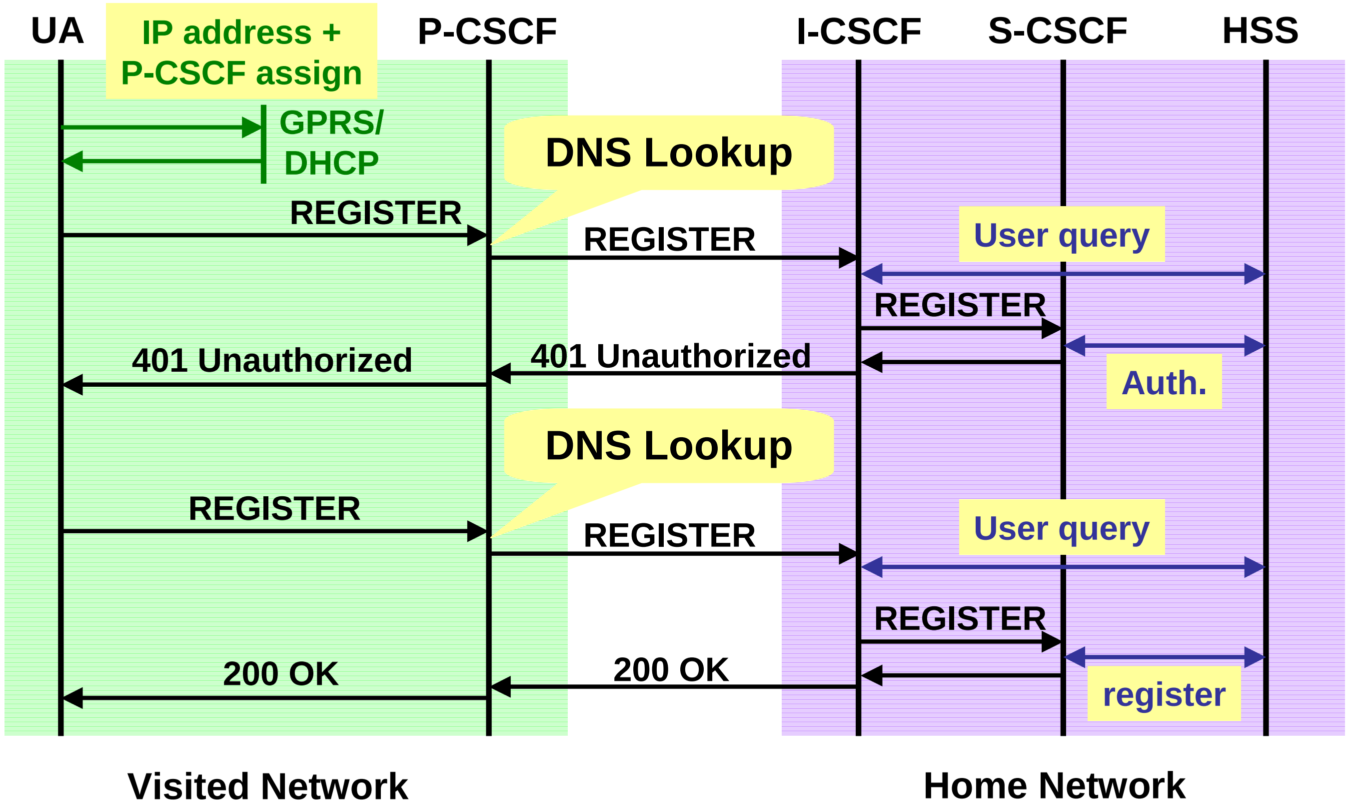
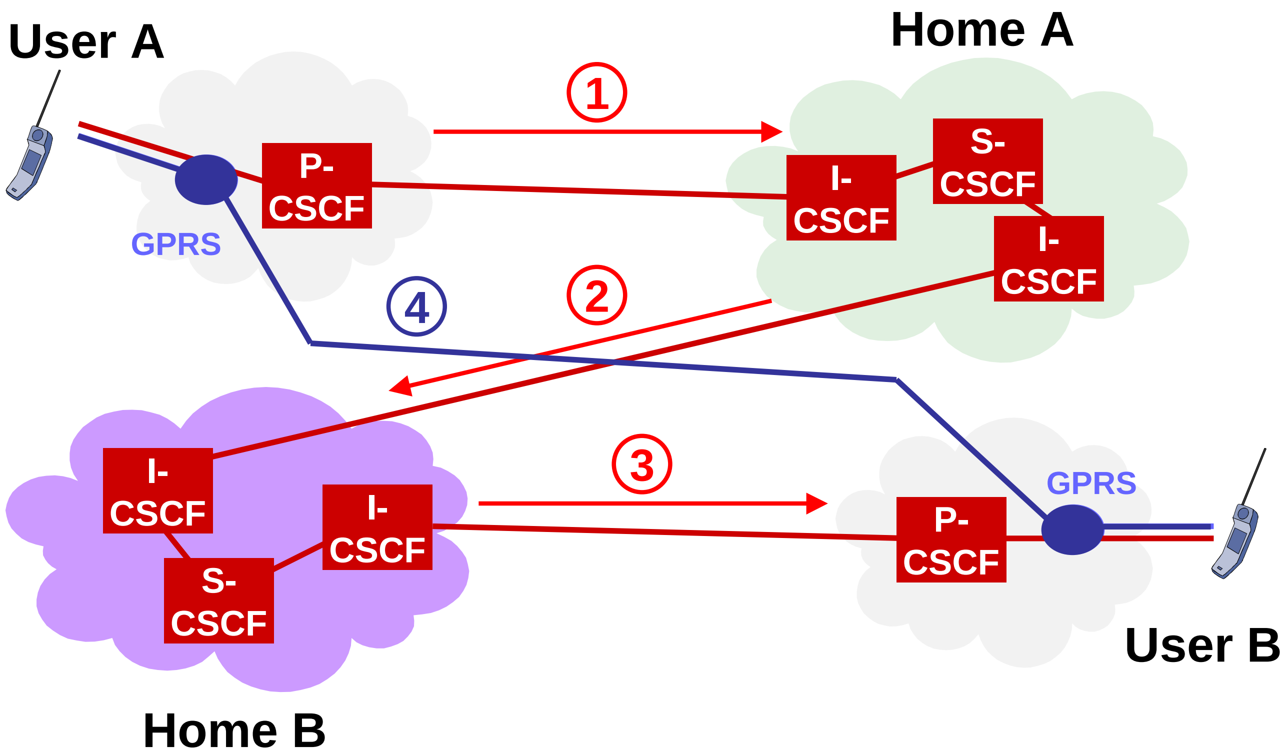
3G Roaming Scenario: Registration



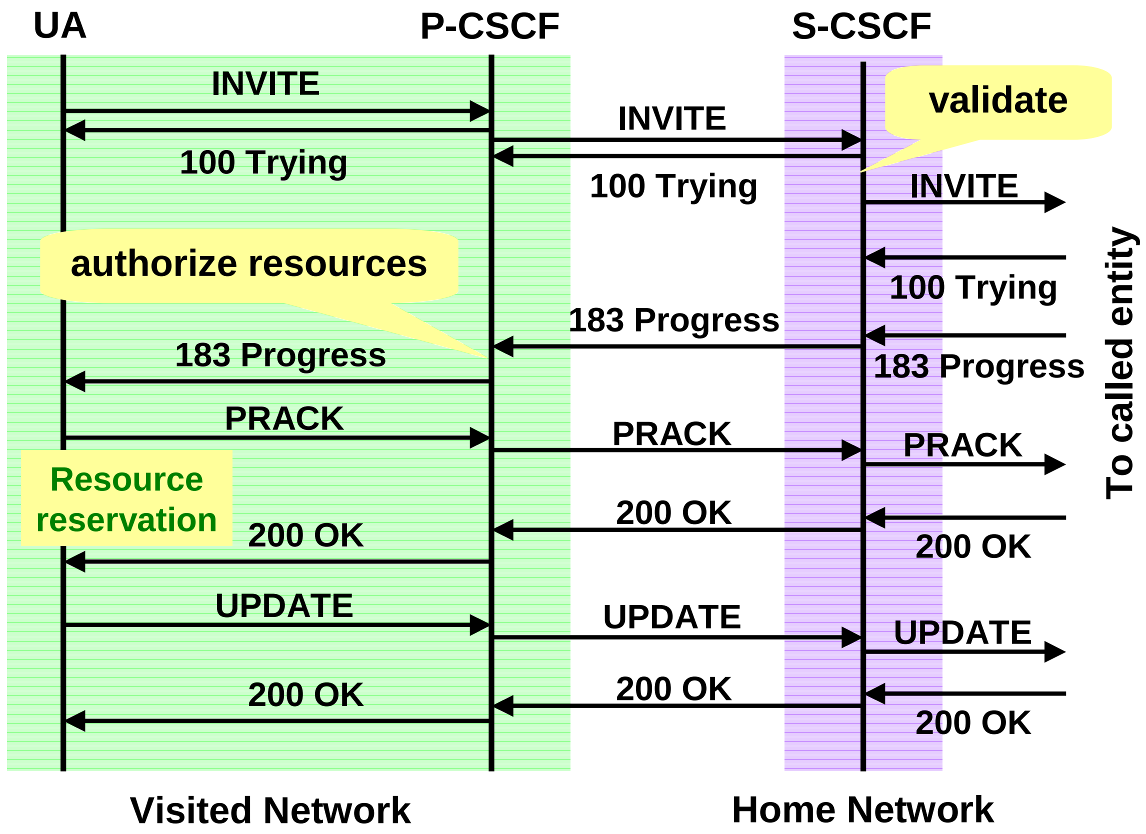
SIP Registration of a Mobile Node



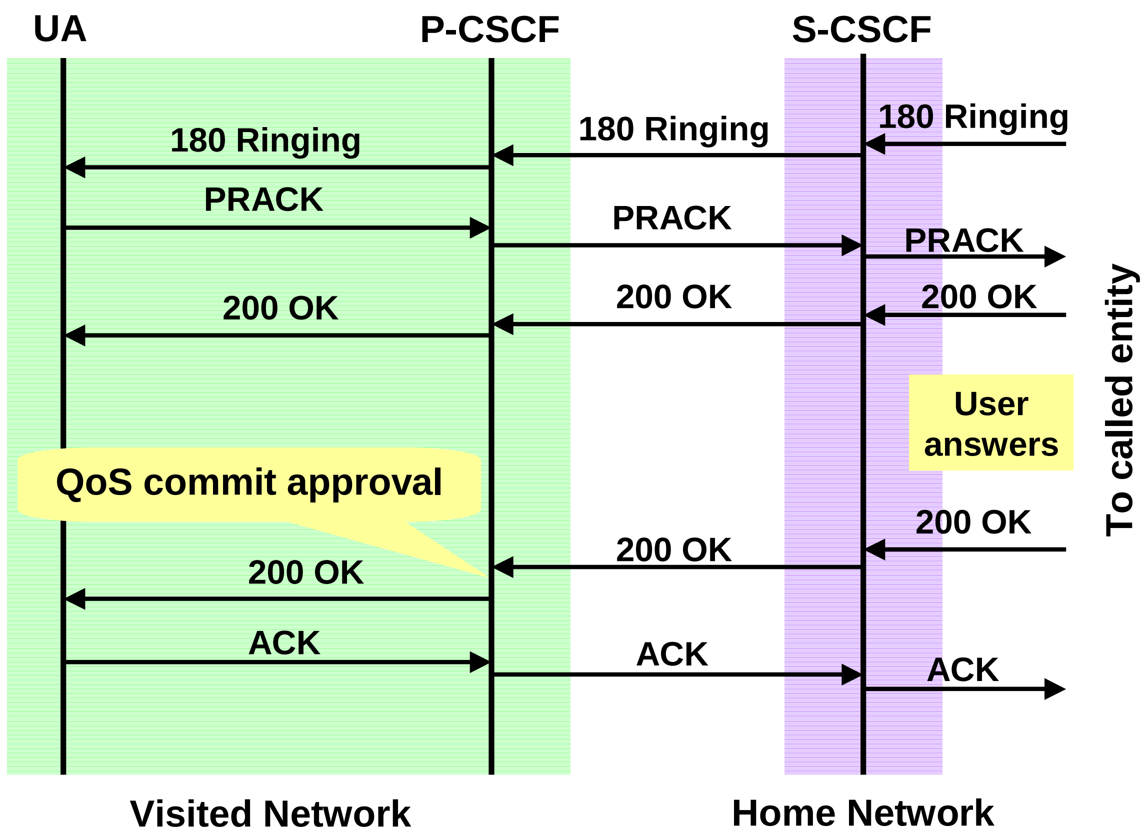
3G Roaming Scenario: Simple Call



Simple SIP Call: Caller Side (1)



Simple SIP Call: Caller Side (2)



Some further 3G Signaling

- Called side: pretty much the same
- Call termination by either endpoint: straightforward
  - Simple two-way handshake: BYE – 200 OK
  - Release associated GRPS resources
  - Accounting in S-CSCF/HSS: e.g. generate Call Detail Record (CDR)
- Call termination by "network"
  - S-CSCF generates SIP BYE requests to called and calling UA
- Terminal de-registration by user
  - Simple two-way SIP handshake: REGISTER – 200 OK
- Terminal de-registration by "network"
  - Uses SIP NOTIFY mechanism to inform the UA

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SIP Features used in 3G

- SIP Base Spec (RFC 3261)
- Reliable Provisional Response (PRACK) (RFC 3262)
- Session Description Protocol (RFC 2327)
- SDP Offer/Answer Scheme (RFC 3264)
- SUBSCRIBE / NOTIFY method (RFC 3265)
- SDP extensions for IPv6 (RFC 3266)
- UPDATE method (RFC 3311)

- Resource reservation extensions ("manyfolks")
- Authentication and privacy

… among others … + … a number of extensions …

Service Routing Extensions in 3G

- P-CSCF and S-CSCF perform central functions for UAs
  - Calls need to pass (at least) through both of them
  - to ensure all services are available
- Configure P-CSCF as outbound proxy
  - Address obtained during link layer initialization
- Record proxy chain to be traversed during registration
  - SIP extension for recording proxies to be traversed en-route
  - Path: header
  - Used together with loose source routing from S-CSCF to UA
- Learn about S-CSCF in REGISTER response
  - SIP extension for service routing
  - Service-Route: header
  - Used together with loose source routing (as defined in RFC 3261)

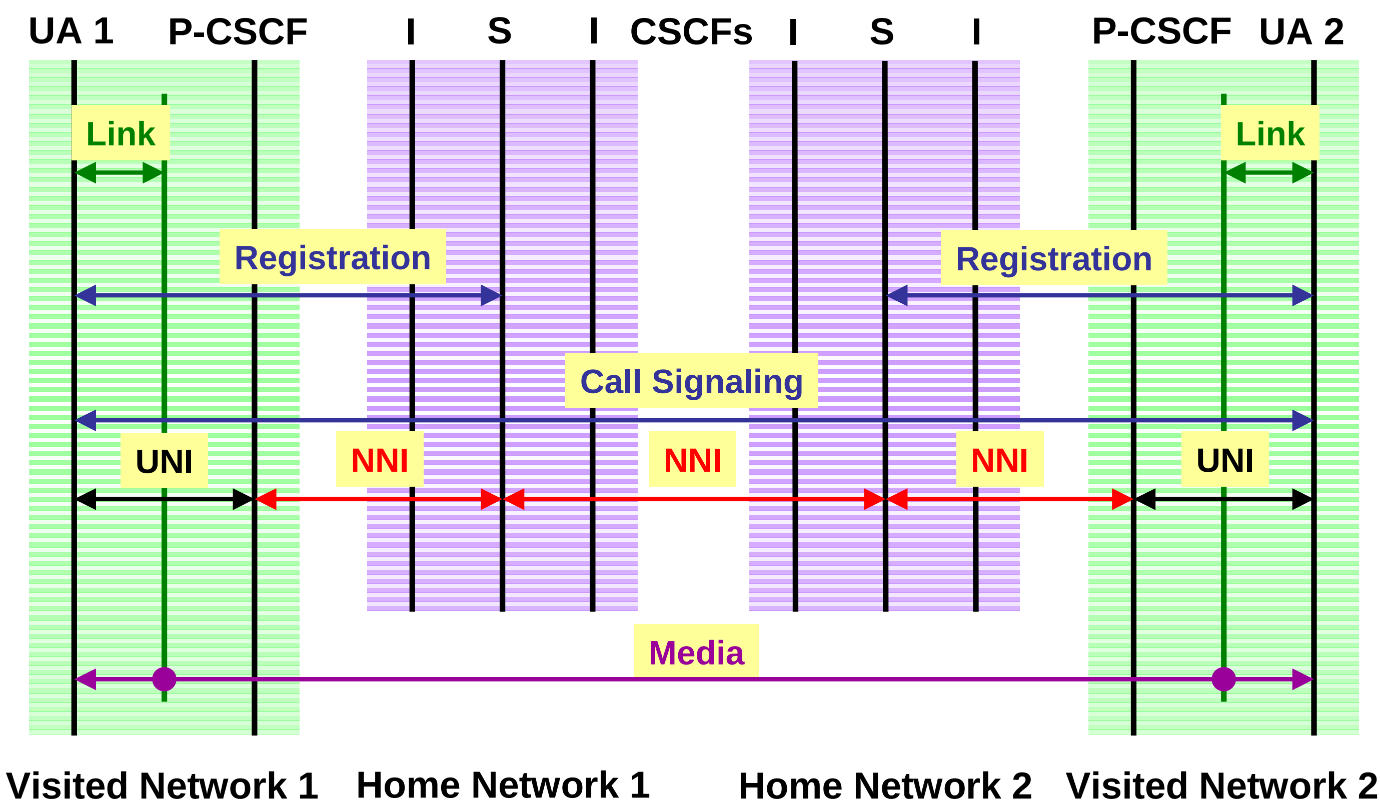
Some SIP Security in 3G

- Basic Approach: closed network infrastructure
  - Transitive trust (and secure communications) within the network
- SIP Digest Authentication for registration
- Stateful network infrastructure
  - Only registered users can place calls
  - Ingress nodes (P-CSCFs) validate UA requests
  - S-CSCF additionally check validity per request
- P-Asserted-Identity: header
  - May contain network-generated (anonymized) identifier
- Privacy: header to select CLIP/CLIR
  - User id may be removed per user request when leaving the network

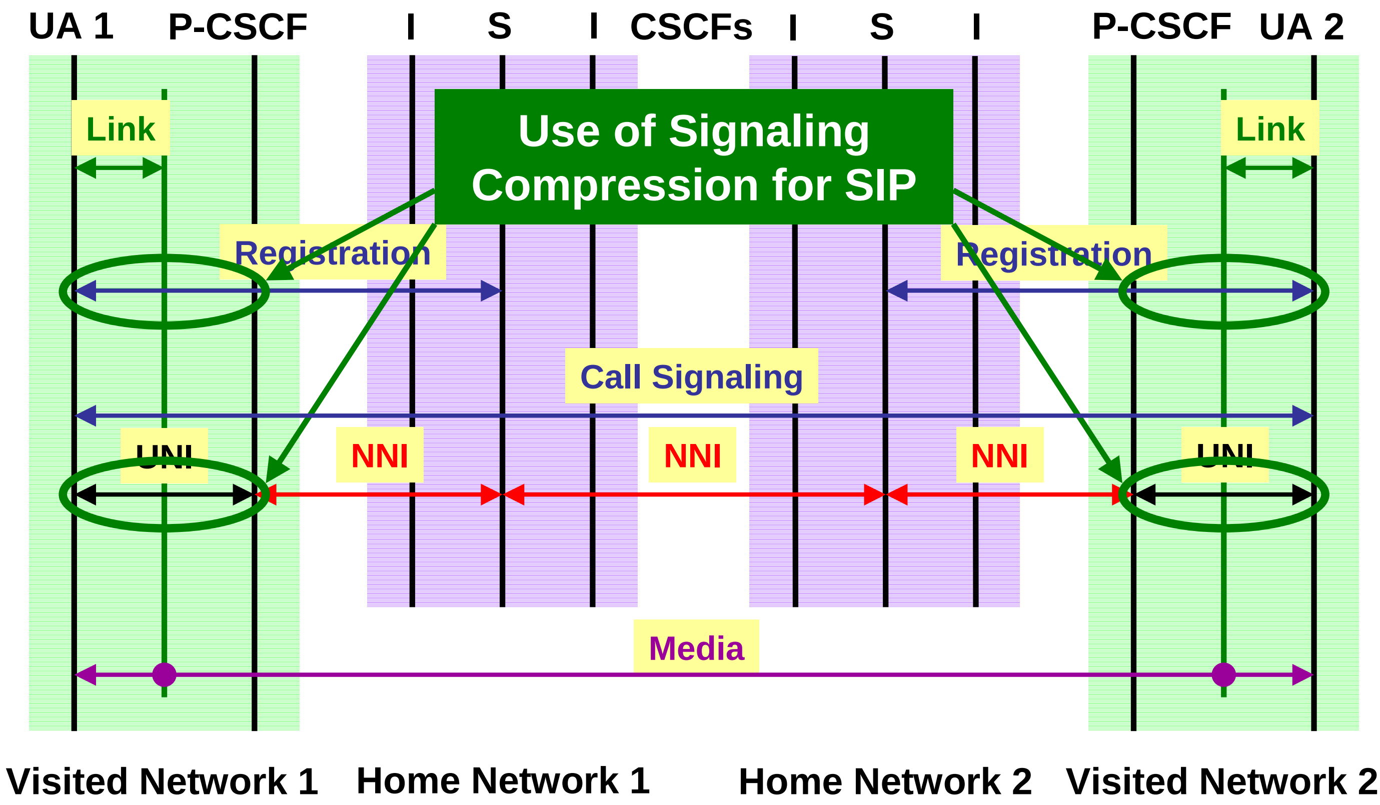
Further 3G Extension Headers

- **P-Associated-URI:**
  - URIs allocated by service provider returned in REGISTER response
- **P-Called-Party-ID:**
  - To indicate original request URI to the called party
- **P-Visited-Network-ID:**
  - Conveys the identifier of the currently visited network
- **P-Access-Network-Info:**
  - Reports information about the currently used access radio link
- **P-Charging-Function-Addresses:**
  - Indicates where to report charging information (e.g. CDRs) to
- **P-Charging-Vector:**
  - Enable correlation of charging information across the network

Summary: Signaling and Media Flows in 3G



Summary: Signaling and Media Flows in 3G



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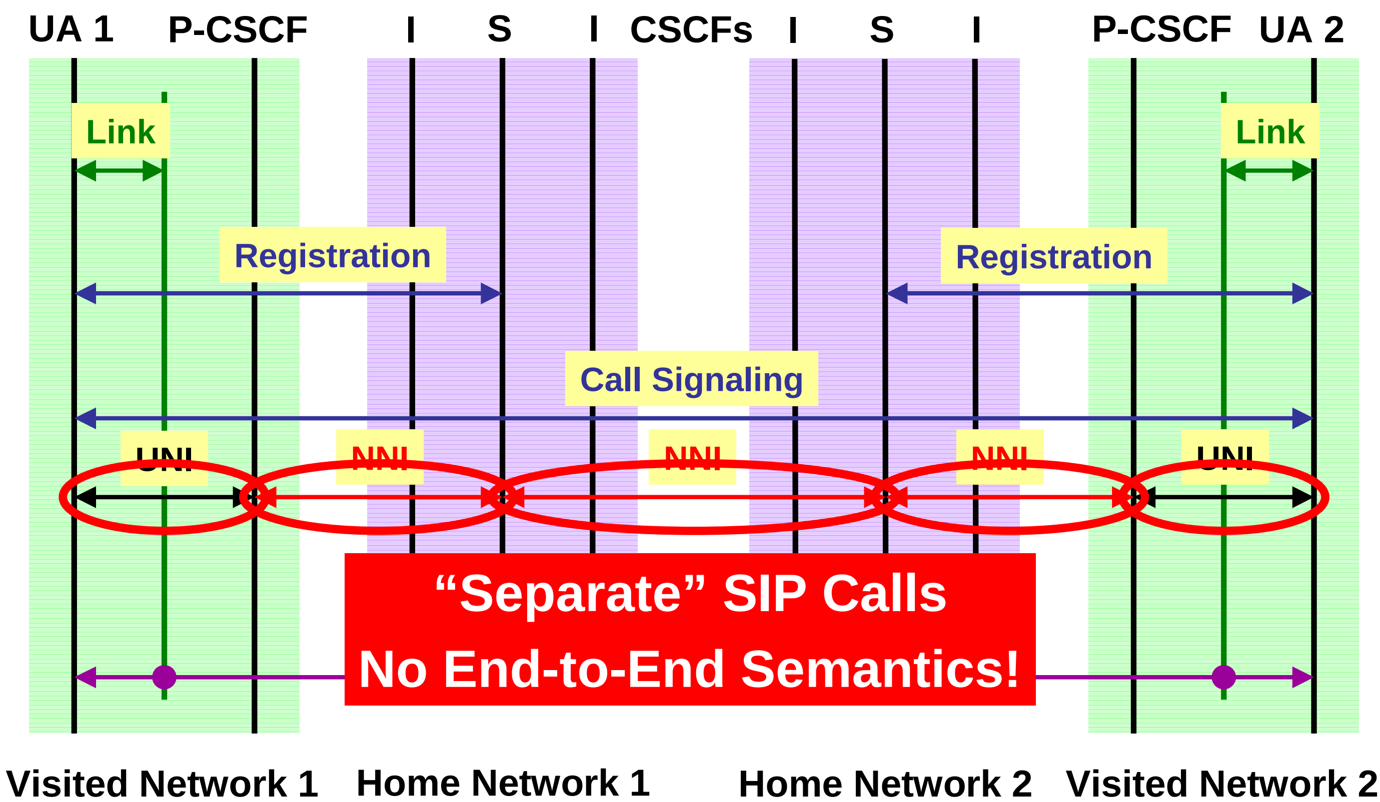
Summary: Signaling and Media Flows in 3G



Service Creation in 3GPP

- Standardized functional building blocks in the network
  - services to be created on top of those
  - Partially standardized services
- "APIs" for service providers
- SIP for interfacing to Application Servers
- Service creation in the network
  - "closed" environment
  - potential for limited outside access
- BUT: No end-to-end signaling
  - Header fields may be removed by P-CSCF
  - New methods may not be passed through
- No / limited end user or third party innovation possible

Conclusion: SIP and 3G

3G is partially built the *old* way – the telephony way…

- Breaks the (S)IP end-to-end model
  - built-in limitation of innovation
- Enforces net-centric service creation
  - built-in protection against newcomers
- Closed architectural model
  - built-in gateways even to external SIP devices

(obvious economic motivations for the above)

- Many boxes, many lines, many interfaces
  - High degree of complexity? (! KISS)

Further Information

http://www.ietf.org/html.charters/sip-charter.html
http://www.ietf.org/html.charters/sipping-charter.html
http://www.ietf.org/html.charters/simple-charter.html
http://www.ietf.org/html.charters/impp-charter.html

http://www.softarmor.com/sipwg
http://www.softarmor.com/sipping

http://www.cs.columbia.edu/~hgs/sip

http://www.3gpp.org/
ftp://ftp.3gpp.org/