Session Initiation Protocol

SIP protocol and its extensions

SIP Service Architecture

SIP in 3G

A lot of this material
is based on proposals =>
may change quickly

Sources

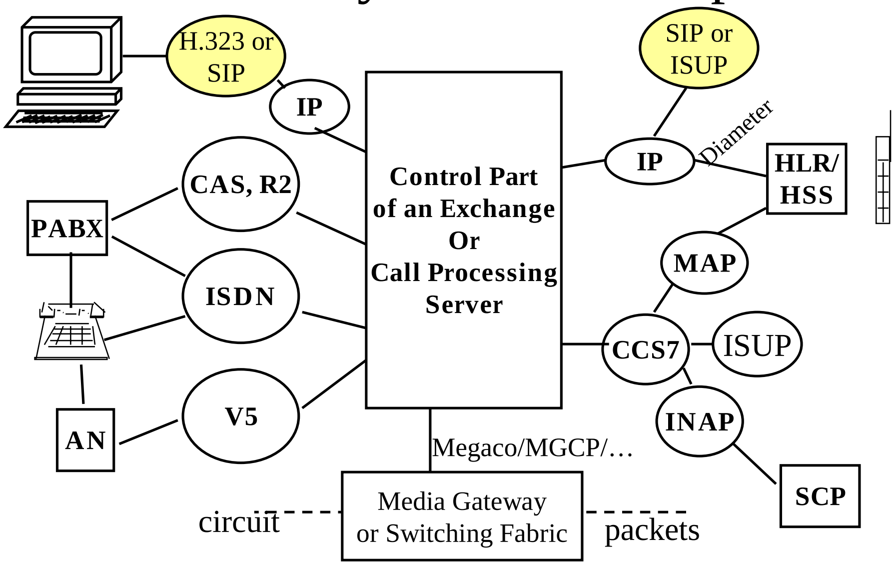
IETF:

RFC 3261: SIP: Session Initiation Protocol
RFC 3262: Reliability of Provisional Responses in SIP (PRACK)
RFC 3265: SIP Specific Event Notification
RFC 3311: SIP UPDATE method
RFC 3398: ISUP to SIP mapping
RFC 3428: SIP Extension for Instant Messaging
RFC 2327: SDP: Session Description Protocol
RFC 3264: An Offer/Answer Model with Session Description Protocol (SDP)

3G Release 5:

3GPP TS 24.228 v5.2.0 (2002-09) Signaling flows for the IP MM call control based on SIP and SDP; stage 3 (Release 5)
3GPP TS 24.229 v5.3.0 (2002-12) IP multimedia call control protocol based on SIP and SDP, Stage 3 (Release 5)
3GPP TS 29.228 v5.1.0 (2002-09) IMS Cx and Dx interfaces, Signaling flows and message contents; (Release 5)
Etc…

Summary of course scope



SIP Requirements and fundamentals

- Part of IETF toolkit
  - Reusing other protocols & mechanisms: HTTP, etc.
  - Flexible
  - Extensible
- Moves intelligence to End System entities
  - End-to-end protocol
- Interoperability
- Scalability (although some state in network)
- Service creation easy
- URLs and Addresses are reused
- Same routing as SMTP
- Reuses infrastructure (all applications will use SIP entities for different services)

SIP overview

- Simplicity
  - Ascii based - simple tools for development
  - Lower call setup time than in H.323
  - basic protocol + extensions structure adopted
- Caller preferences, Ability to support many media types
- Runs over UDP or TCP (or SCTP)
- Used between both service and call control entities
- Has been adopted for 3G IP Multimedia signaling
- Originally subscriber signaling, proposed also as network to network signaling
- A lot of development during the last 3…4 years!

SIP Implementation Status of 11/2003

There are several single-operator single-vendor islands offering SIP services, but

The inter-operator or multi-vendor solutions are still very rare, and need to be designed case by case.

This despite the Interoperability Bakeoffs organised by vendors!

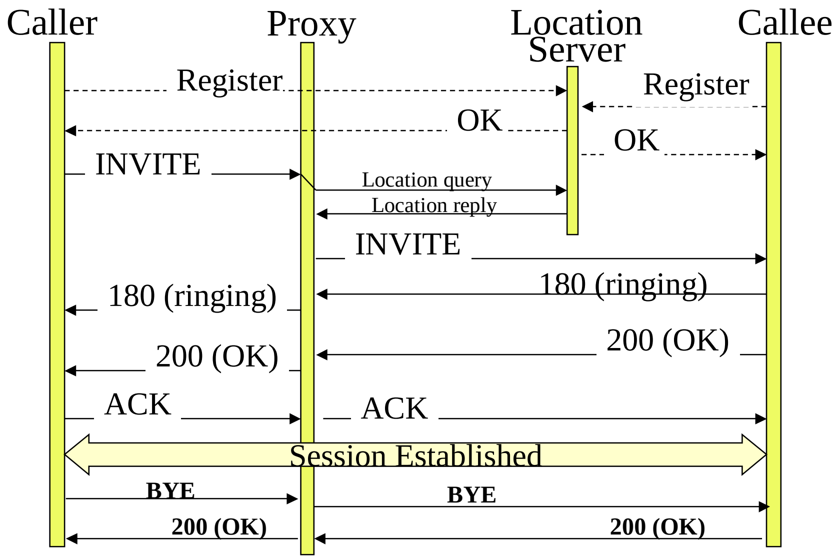
SigComp allows cmpression of Signaling Messages

- RFC 3320 and RFC 3321 specify a layer between the signaling transport and the signaling application
- Uses Global and User Specific Dictionaries to store state data over many SIP sessions
- Overall Compression/decompression architecture is based on a bytecode driven Universal Decompression Virtual Machine
  – Bytecode can be sent in SigComp messages by the Compressior
  – leaves a lot of detail for the implementor

Sip Entities

- User Agents
  – Can act as client and as server
- Servers:
  – Redirect Servers
    - Send back alternative location of the user (similar as HTTP servers)
  – Proxy servers
    - Act on behalf of client (forwards requests)
    - Forking proxies
  – Registrars
    - Accepts registrations
  – Location Servers (not part of SIP architecture)
    - Gives back location of user (received from registrars)
    - E.g. HSS in 3GPP IMS architecture
    - Protocol between Location server and SIP server <u>not defined by SIP specs</u> (e.g. LDAP)

Basic SIP call setup and release



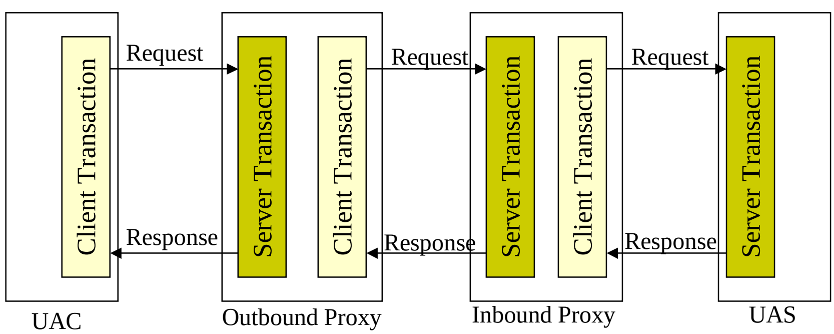
”Basic call” Example

- Caller sends INVITE
- Callee can accept, reject, forward the call
- If the callee accepts the call: responds with an optional provisional (1xx), and a final (≥200) response
- The caller confirms final response via ACK
- Conversation
- Caller or callee sends BYE
- BYE is acknowledged by 200 OK
- Low call setup times, post dial delay: 1.5 RTT !

SIP messages have headers and a body

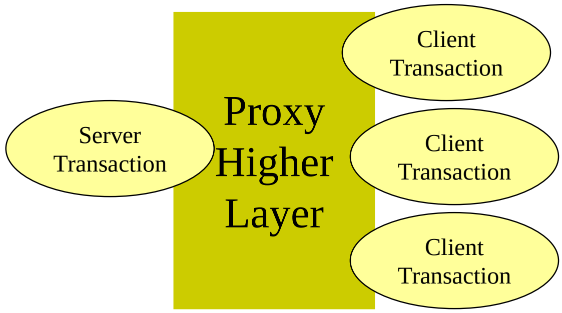
- Headers carry control information and are processed e.g. by Proxies
- Body can be e.g. SDP – session description protocol
  - end-to-end information (cmp H.245) describing session requirements e.g. coding methods, etc
- Message delivery is transaction oriented= have request + reply: e.g INVITE+200 OK

User Agent is split into User Agent Client (UAC) and User Agent Server(UAS)



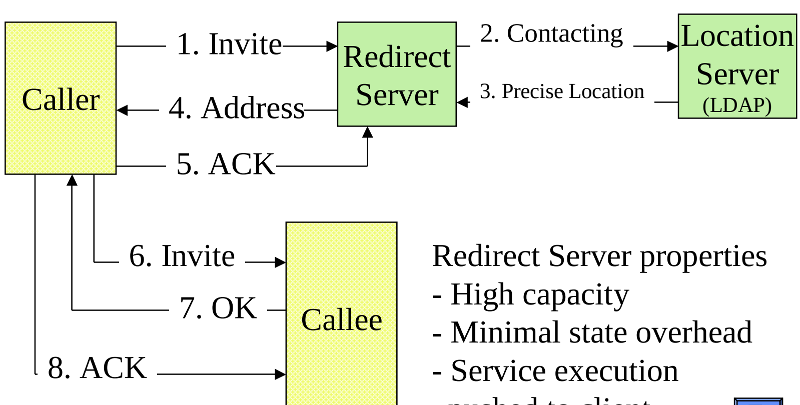
- All Communication follows this transaction model, except 2xx and ACK.
- Server transactions filter incoming requests, absorbing retransmissions
- Only UAS can generate 2xx and only UAC can generate ACK (to success).

A Stateful Proxy can fork a transaction



Forking = multicast of INVITEs to N addresses

Redirect Server pushes processing to clients



Redirect Server properties
- High capacity
- Minimal state overhead
- Service execution pushed to client

17.3.04

Stateful Proxy vs Stateless Proxy

- Maintains call context
- Replicates UAS/UAC to process requests and responses
- Call state and transaction state can be maintained
- Forking proxies require state
- TCP proxies must be stateful for reliability
- Enhanced services require state
- Can collect charging info

- No call context
- Response is not based on UA replication
- Provides client anonymity
- Restricted gateway access
- High processing capacity
- Easier to replicate than the stateful proxy
- Also semi-stateful is possible

UA = User Agent, UAC = UA Client
UAS = UA Server

Notes on SIP methods: INVITE

- Requests users to participate in a session. Body contains description of the session. If someone wants to modify the parameters of the session, he/she must re-INVITE carrying the modified parameters.
- One or more Provisional and one Final response are expected.

Notes on SIP methods: ACK

- Acknowledges the Final Response to INVITE even if INVITE was cancelled → result is 3-way handshake: INVITE-final-resp – ACK.
- Proxies can only ACK non-successful Final Resp.
- Purpose:
  - Let’s the server know that session establishment was successful.
  - Forking may result in many final responses. Sending ACKs to every destination that sent a final response is essential to ensure working over UDP.
  - Allows sending INVITEs without session description. In this case the description is postponed to ACK.
- Has the same Cseq as the INVITE it acknowledges (see later for SIP headers).

Notes on SIP methods: CANCEL

- Purpose: to cancel pending transactions. Will be ignored by completed transactions = final response already sent.
  - useful for forking proxies. If one destination answered, the forking proxy can cancel all other pending INVITEs.
- Has the same Cseq as the request it calcels (see later for SIP headers).

Notes on SIP methods: REGISTER

- Purpose: to register the user’s current location.
- A user can be registered in several locations at the same time. Forking is used to find out where the user wants to answer the session invitation.
- A user can register from anywhere to his registrar → provides mobility.

Notes on SIP methods: OPTIONS

- UA can query a server: which
  - methods and
  - extensions and
  - which session description protocols it supports.
  - which encoding for message bodies the server understands (e.g. compression to save bw).

Some SIP issues

- Parties can release the “call session” but since they have obtained each others IP-addresses, they can continue sending media streams to each other!!
- How to push INVITE to B-party, if B-party does not have a permanent IP address which is most often the case!

Integration of Proxy with Firewall and NAT

- Response messages (e.g. 180) are not reliably delivered. This may cause tear down of the call if it was initiated from ISDN

PRACK method

- If BYE is lost, Proxy does not know that call has ended

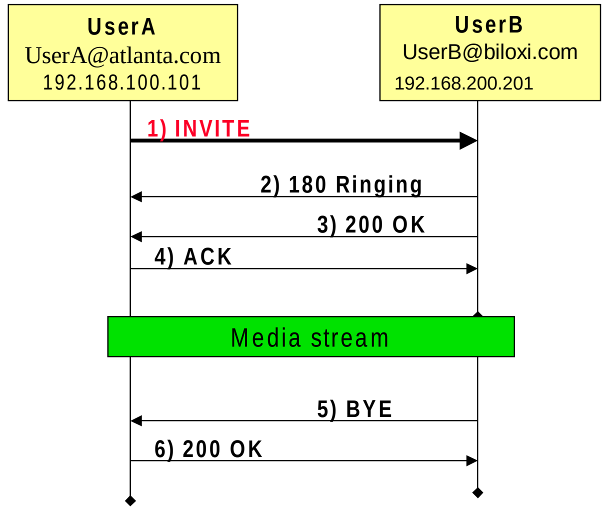
KeepAlive = re-INVITE mechanism

- Ascii coding increases the signaling overhead in Radio access

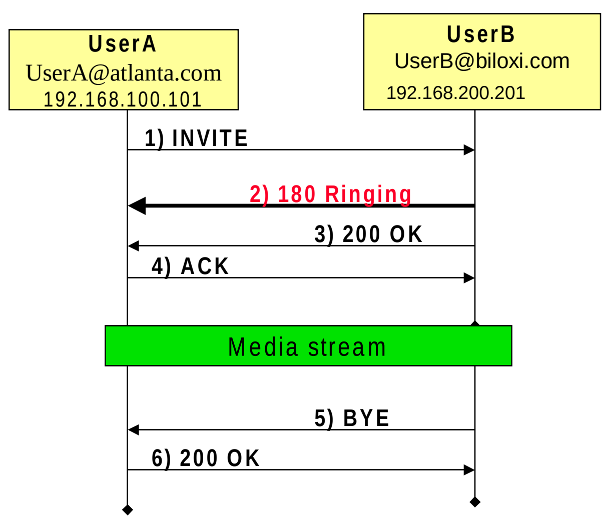
Addressing

- **sip:user@host[parameters][headers]**
- SIP-addresses are like URLs, with prefix sip: which gives schema
  - sip:joe.smith@hut.fi
  - sip:joe.smith@hut.fi?subject=Protocol
  - sip:sales@hotel.xy;geo.position:=48.54_-123.84_120
- Address must include host, other parameters are optional (username, port, etc…)
- Email-addresses can be reused
- “Click-to-call” on web-pages, MM messages, etc… are easily implemented

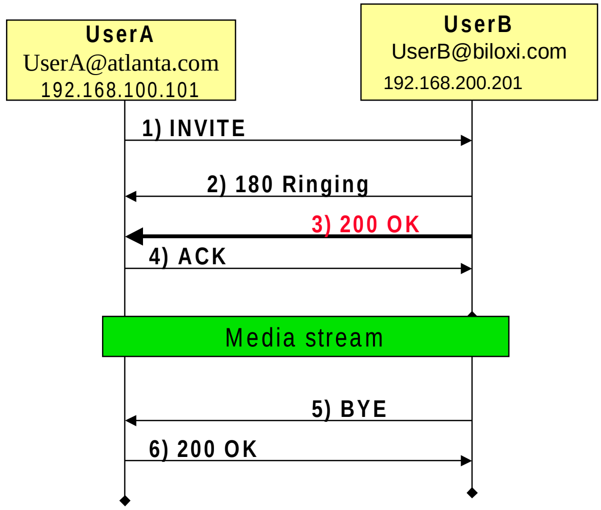
”Basic Call” call flow



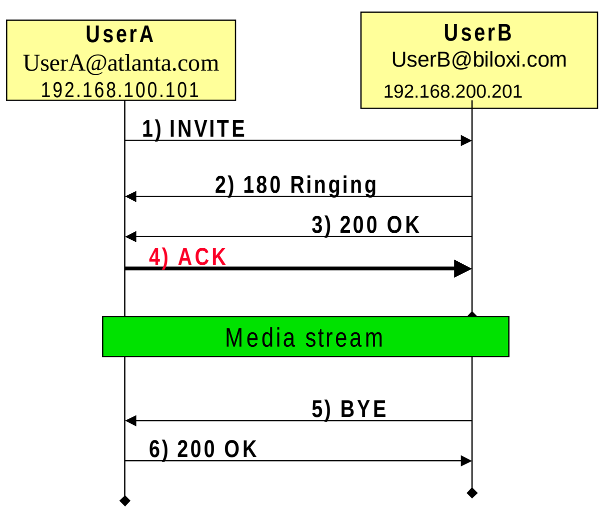
”Basic Call” call flow



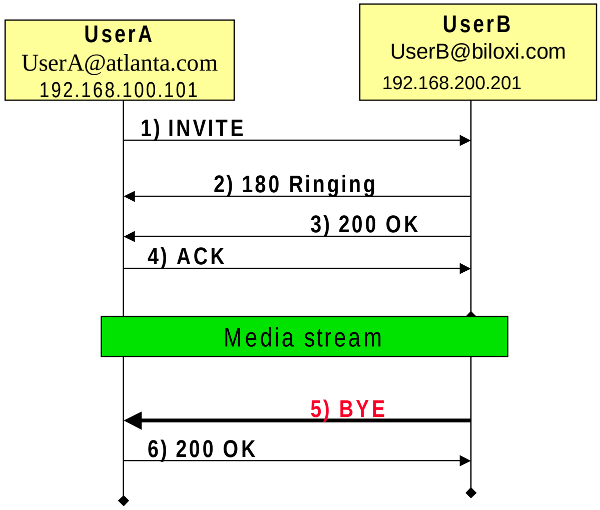
”Basic Call” call flow



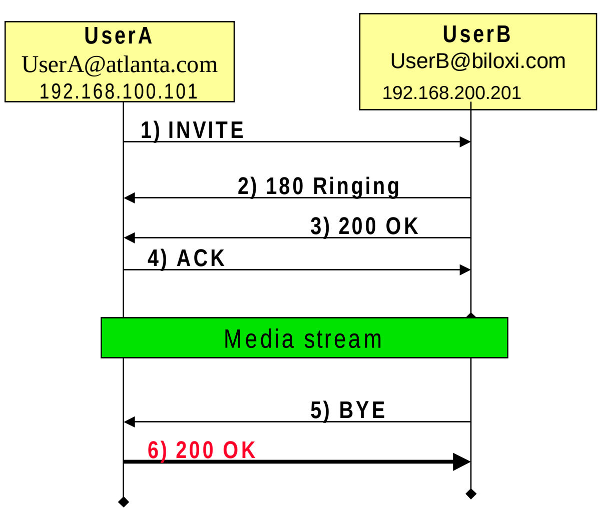
”Basic Call” call flow



”Basic Call” call flow



”Basic Call” call flow



Requests invoke SIP methods

- SIP methods are invoked on servers when requests arrive:
  - A REGISTER request sends location information of users to Registrars, registers with the location service
  - An INVITE request invites a user to participate in a session or conference
    - The message body contains a description of the session (usually SDP)
  - ACK requests are used to confirm responses for INVITE, for reliable message exchanges
  - CANCEL requests cancel the pending request of the session
  - BYE requests are used to terminate active sessions
    - Any party of the session can send it
  - OPTIONS requests are used to query information about servers' capabilities
  - PRACK requests are used to confirm provisional responses

SIP responses are classified by first digit

- HTTP look-alike
- Hierarchically organized three digit codes: status code - text associated with the code
- Provisional and final responses:
  - 1xx responses are informational messages e.g., 180 Ringing
  - 2xx response shows a successful transaction e.g., 200 OK
  - 3xx responses are redirect messages e.g., 301 Moved Permanently
  - 4xx responses indicate errors in requests e.g., 400 Bad Request
  - 5xx responses indicate server errors e.g., 500 Version not supported
  - 6xx responses indicate global failures e.g., 600 Busy everywhere

SIP Message Format

- START-LINE
  - SIP version used
  - In requests: address and method used
  - In responses: status code
- HEADERS
  - Information about call
- BODY (payload)
  - Usually SDP message

```
C->S: INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Start line

Headers

Body

To and From header fields

- **To: specifies the logical call destination**
- **From: specifies the logical call source**
- **Present in all SIP messages**

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```

Call-ID and CSeq header fields

- **Call-ID: It helps to uniquely identify a particular SIP dialog or registration**
  - It helps to match requests and responses
  - It helps to detect duplicated messages
- **CSeq: It is a number that uniquely identifies the transaction in a call**
- **Present in all SIP messages**

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```


Content-Type and Content-Length header fields

- **Content-Type: It describes the media type of the message body**
- **Content-Length: The number of octets in the message body**
  - It is mandatory in all SIP messages.

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```

Max-Forwards

- **Max-Forwards field must be used with any SIP method**
- **It limits the number for proxies or gateways on the way of SIP message to the destination.**

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```


VIA header indicates path taken by the request so far

- **Branch parameter is used to detect loops**
- **Contains transport protocol, client's host name and possibly port number, and can contain other parameters**

```
INVITE sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP server10.biloxi.com;branch=z9hG4bK4b43c2ff8.1
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77ef4c2312983.1
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
Max-Forwards: 68
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```

Record-route and Route

- **Record-Route: header is added by proxy, when proxy wants to stay in the route of all sip messaging**
- **Route is added by User Agent Client, after response came, with all Record-route headers in it (then UAC knows which relays want to stay on the signaling path)**
- **NOT the same as Via: headers**

```
INVITE sip:callee@u2.domain.com SIP/2.0
Contact: sip:caller@u1.example.com
Record-Route: <sip:p2.domain.com;lr>
Record-Route: <sip:p1.example.com;lr>
```

Inserted by proxies p1.example.com and p2.example.com.

```
BYE sip:callee@u2.domain.com SIP/2.0
Route: <sip:p1.example.com;lr>,<sip:p2.domain.com;lr>
```

UA can specify through which proxies this message must go

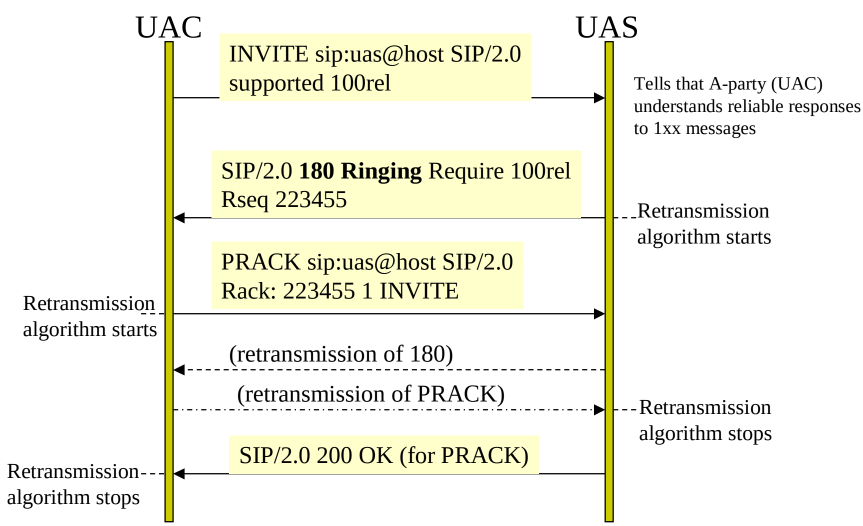
SIP Extensions

- Needed to satisfy additional requirements
- Must conform to design rules
- SIP is not intended to solve every problem (another protocol might be used instead)

Feature Negotiation (OPTIONS)

- *Supported* features can be specified in request and response
  - **Supported** UAC and UAS tell features they support
- *Required* features can be specified in request and response
  - **Require** UAC tells UAS about required options
  - **Proxy-Require** required options for proxy/redirect servers
  - Many extensions use Require and Proxy-Require to specify their support
- New methods can be added without changing the protocol
  - server can respond with **405 Not Supported**
  - returns list of supported methods in **Allow** header
  - client can ask which methods are supported using OPTIONS

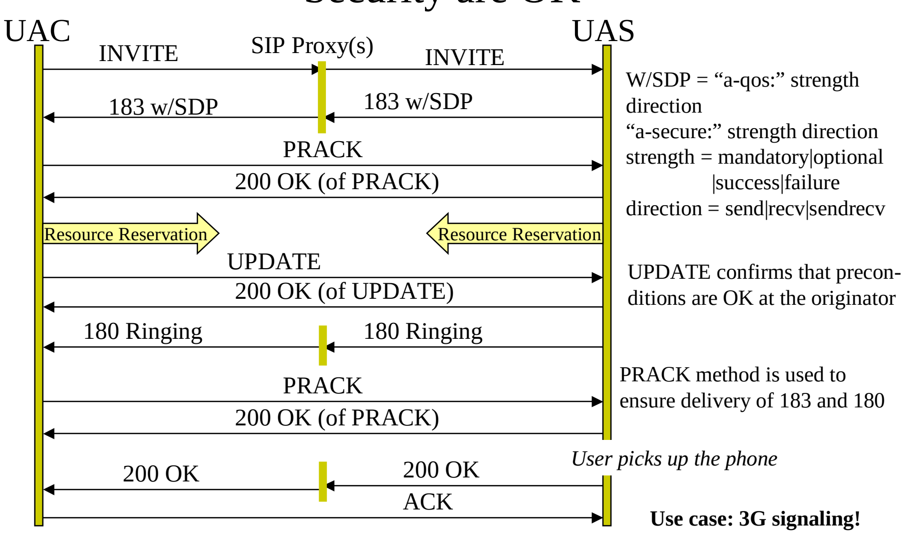
Reliable Provisional response in SIP



QoS support - UPDATE

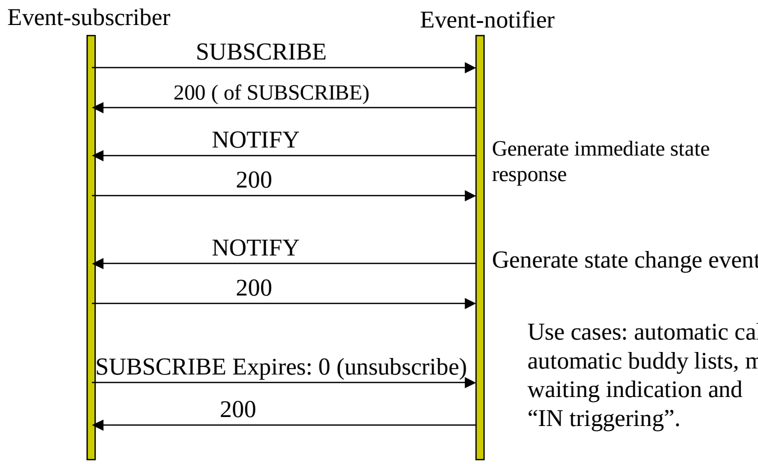
- Usage rule for 183-Session-Progress
  - If “a=qos” appeared in SDP, UAS sends 183 with “Session: qos” and SDP
- Additional Method - UPDATE
  - If “a=qos” appeared in SDP with “confirm” attribute, UAS/UAC sends UPDATE with success/failure status of each precondition.
  - 200 OK must acknowledge the UPDATE message
- Additional Status Response - 580 Precondition Failure
  - If a mandatory precondition can’t be met, UAS terminates INVITE with this status response

Phone should not ring before QoS and Security are OK



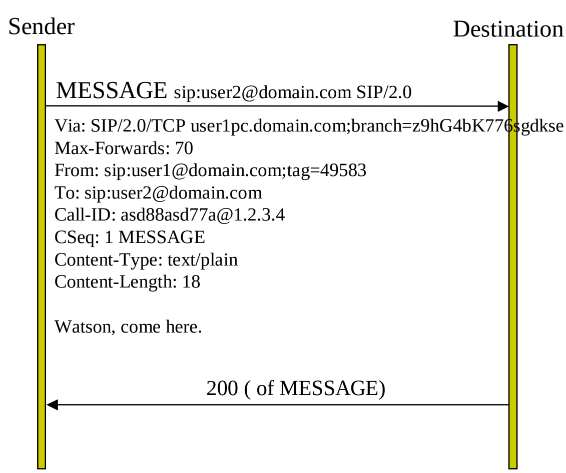
SDP = Session Description Protocol (carried in SIP message body)

SIP event notifications tell about remote significant events to the local party



Use cases: automatic call-back, automatic buddy lists, message waiting indication and "IN triggering".

SIP MESSAGE provides Instant Messaging capability in Pager mode



Note:
-body is MIME or S/MIME
- there is no dialog!

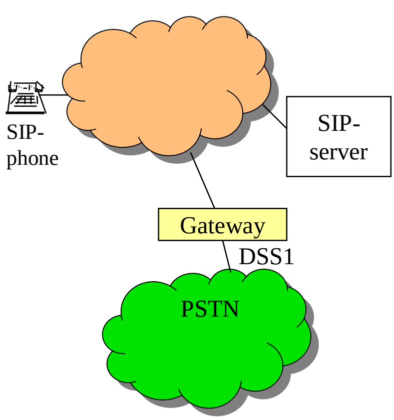
Use example:
-maps to SMS in GSM

Issue:
- authentication of sender and charging! Must use Proper security features

More SIP extensions

- MESSAGE
  - For instant messaging
- INFO
  - To transport mid-session information (very useful in SIP-PSTN gateways to carry all PSTN messages across SIP domains such that do not easily map to any other known SIP message)
- Automatic configuration
  - DHCP or Service Location Protocol (SLP)
- Caller Preferences
  - New headers: Accept-Contact, Reject-Contact, Request-Disposition (e.g. to express a preference for contacting the user at “fixed”, “business” connection).
- REFER
  - For session transfer (Refer-To: and Referred-By:)
- …

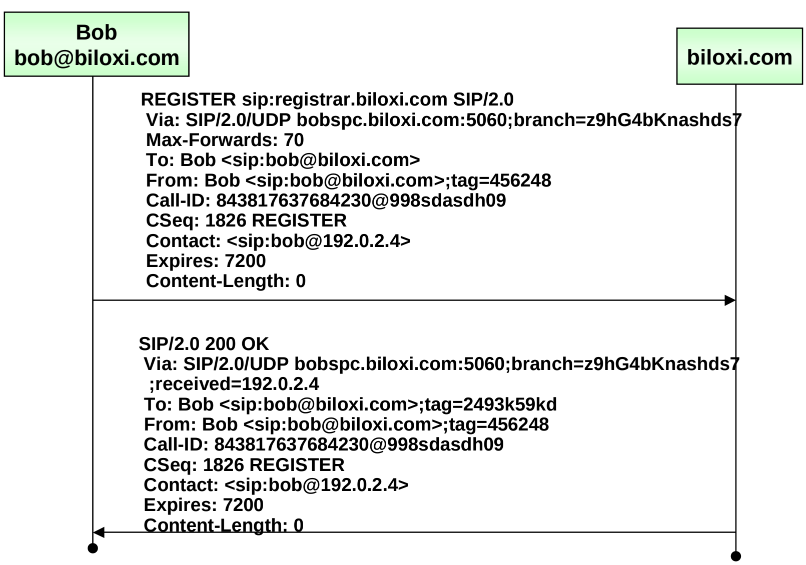
Deployment example: Elisa’s experimental service for BB customers



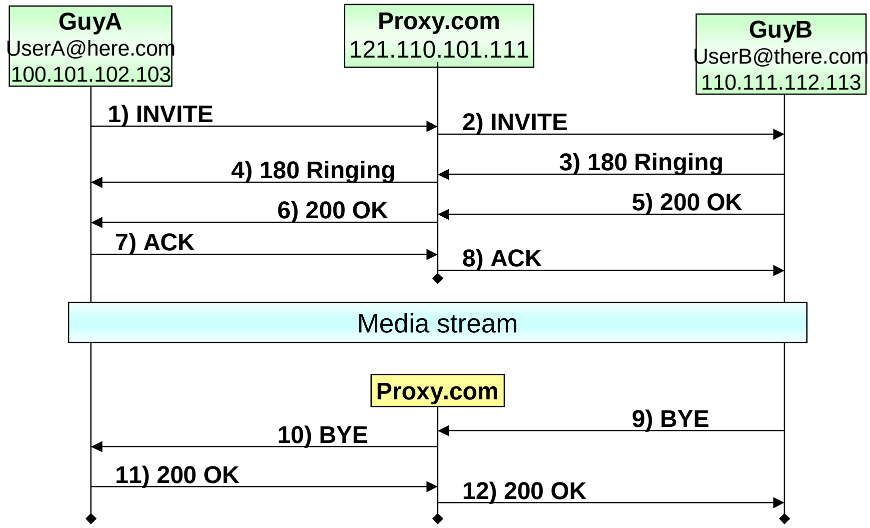
- SIP –server recognizes a numbering block, connects calls directly from IP-phone to IP-phone n the block
- Calls to all other numbers are routed to the gateway
- = SIP-server+Gateway are like a PBX

Call Setup Examples based on Generic SIP

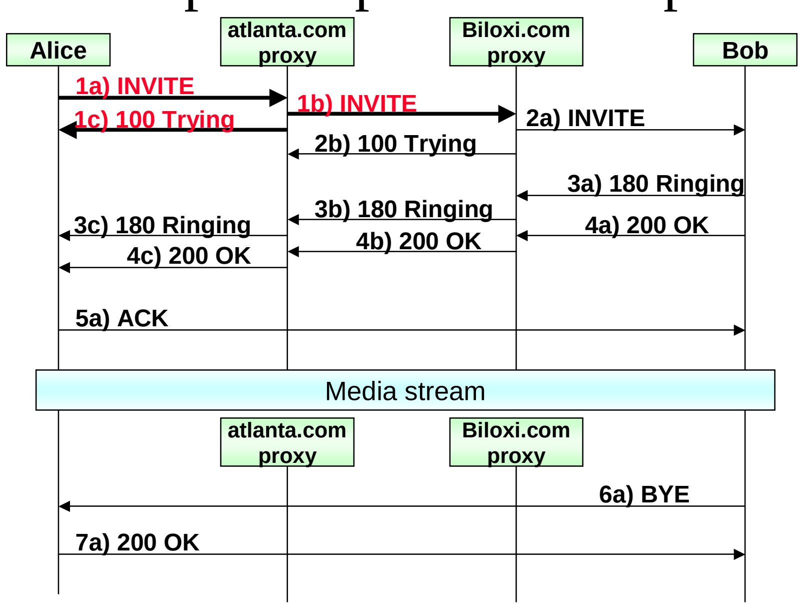
Registration example with SIP



Call Setup example with one proxy



Call Setup example with two proxies



Call Setup example with two proxies

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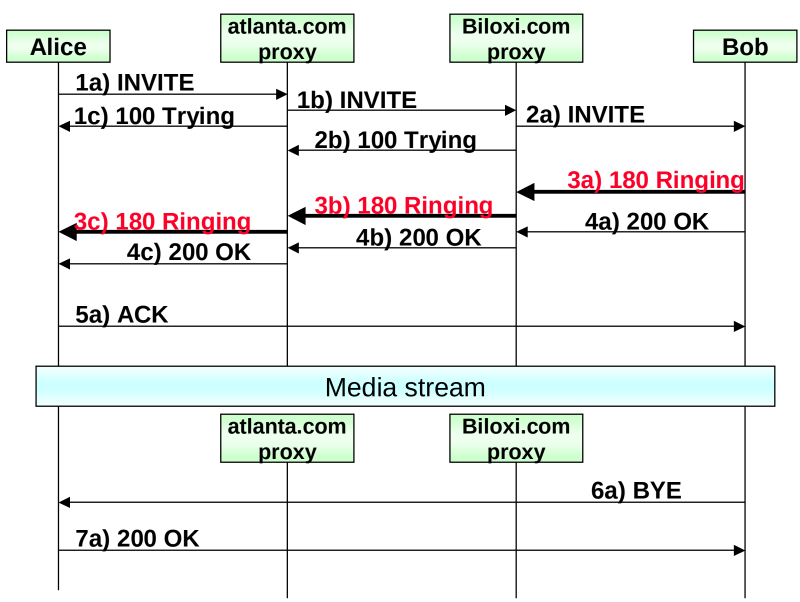

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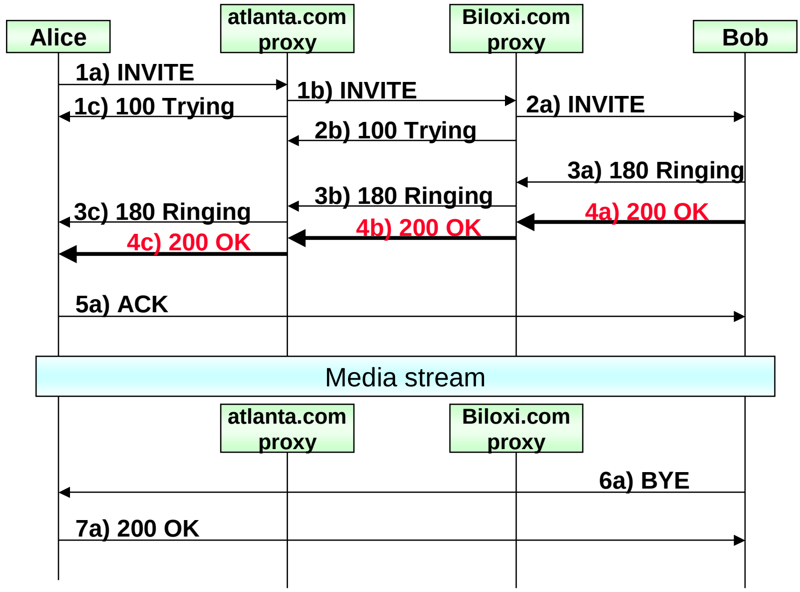

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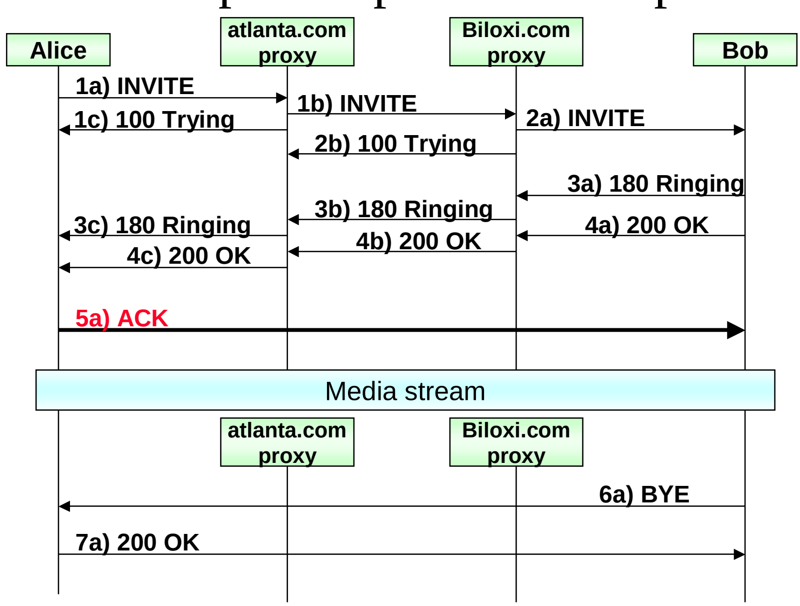

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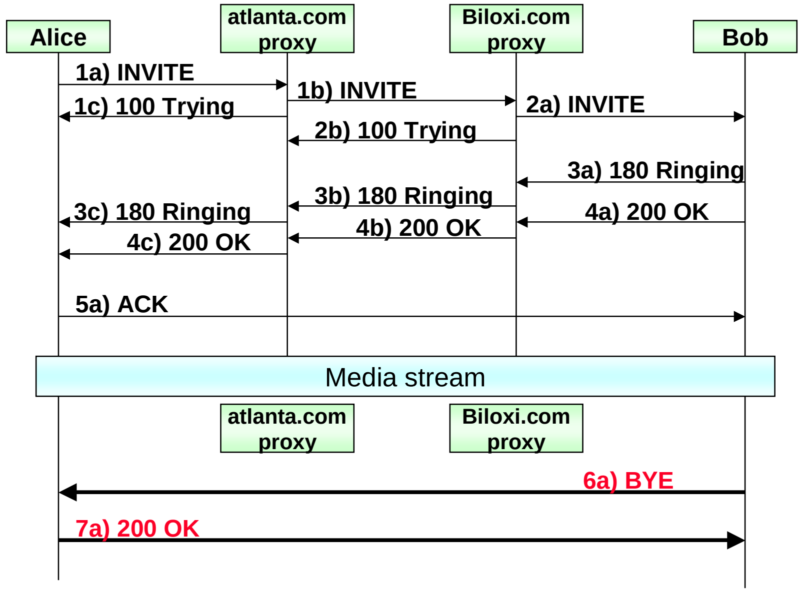
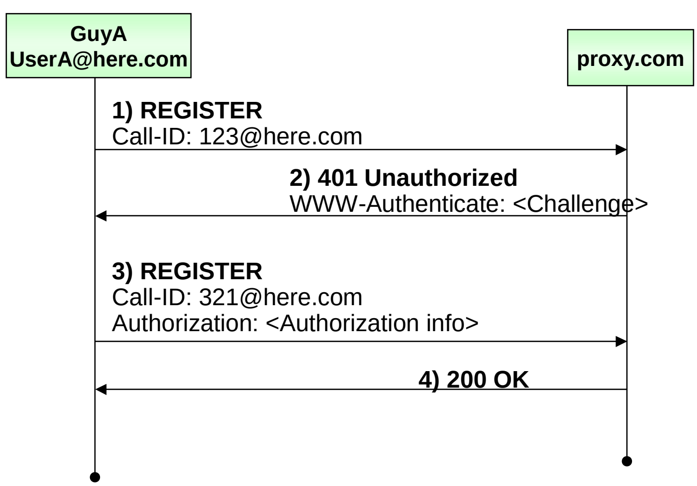
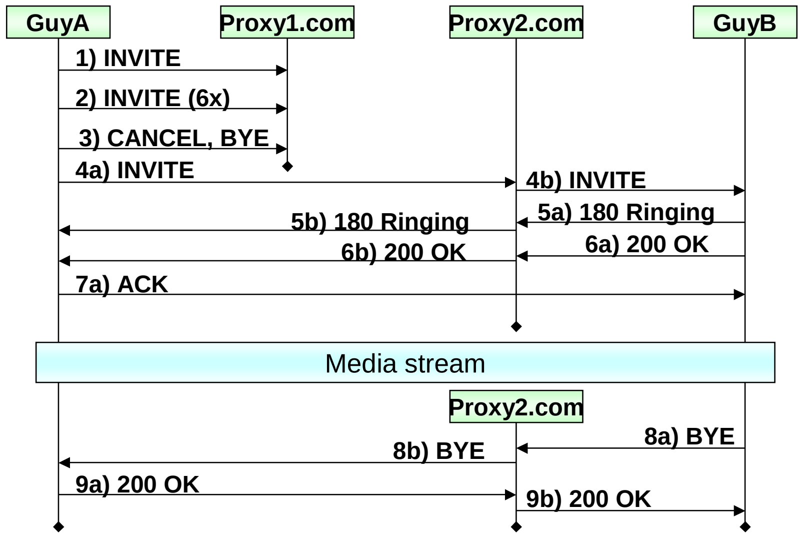


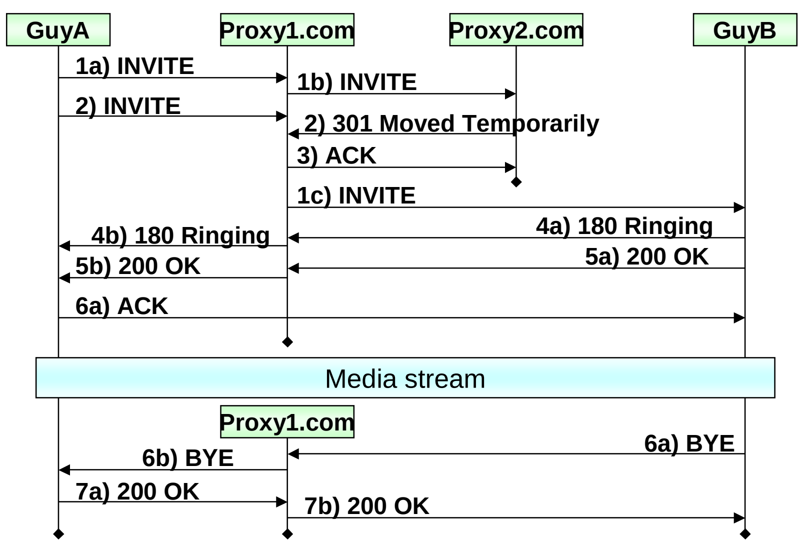
Registration example with SIP authentication



Call Setup example with a non-working proxy



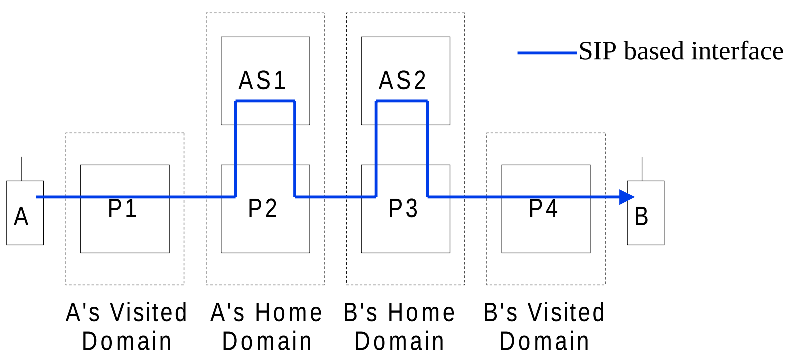
Call Setup example with a Redirect server



Services use many protocols

- New services and more flexible service creation should differentiate IP Communications Network from PSTN
- Services should combine different forms of communication, thus multiple protocols are needed:
  - SIP for media sessions and session related services, subscriptions and notifications?, messaging?
  - HTTP for web and transactions
  - SMTP for e-mail
  - RTSP for media streaming
- The use of these protocols is orchestrated by the service logic: context is set up using SIP.

Routing and Service Model in 3G



P1, P4: Outbound Proxies

P2, P3: Registrar Proxies

AS1, AS2: Application Servers

NB: Also AS based on direct processing of call state: There is no Basic call state model like in IN

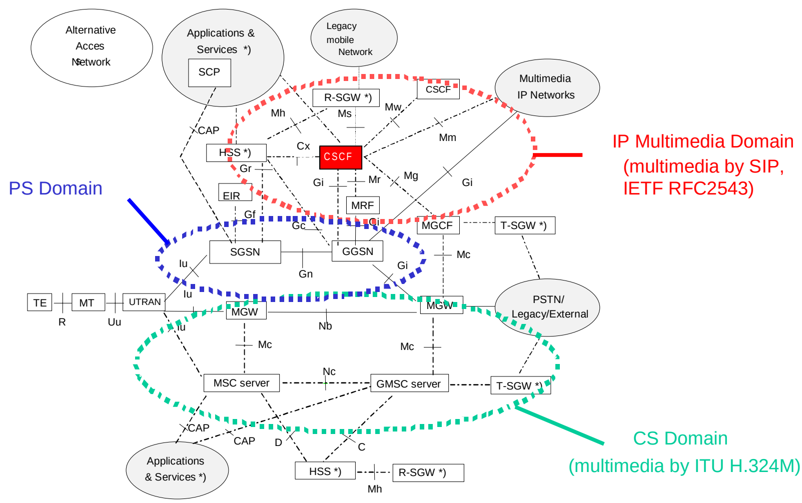
SIP Entities & Service Capabilities

- User Agent (= UAC + UAS)
  - Can run services, such as forwarding, filtering etc.
  - Not always connected (out of coverage/battery etc.)
- Redirect Server
  - Can do services that require only Request-URI change, e.g. translation, parameter addition etc.
- Proxy Server
  - Can change certain headers and stay in the signaling path
  - Forking, actions based on responses
- Back-to-Back User Agent (=both ways User Agent)
  - Can e.g. issue requests to a call leg or modify SDP, generate ACK and 200OK, like UAC/UAS
  - In many cases necessary

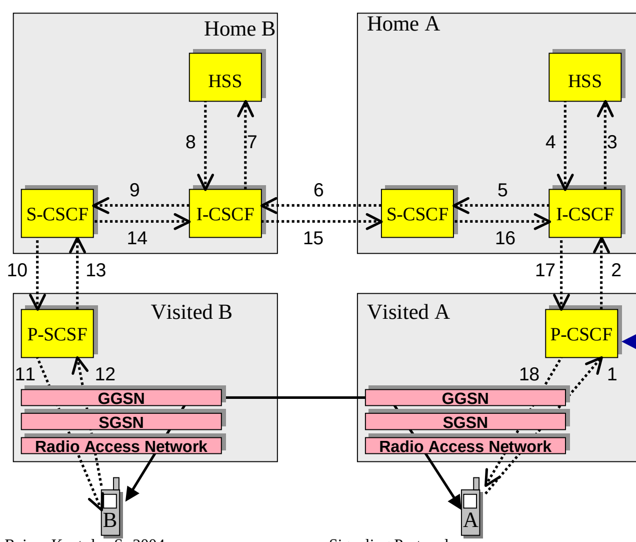
Application Server in 3G?

- Fuzzy Definition but has SIP+ interface!
- Can be a Redirect or Proxy Server or Back-to-Back UA
- The key is that it should be *programmable*
  - Routing based on service logic: what to do when user not registered or busy
  - URI translation: Reachability chains
  - Interfaces to other protocols: HTTP, SMTP, RTSP etc.
- Can be single purpose boxes, or multi-purpose boxes, or controllers who orchestrate things

3GPP Network Model (preliminary: …

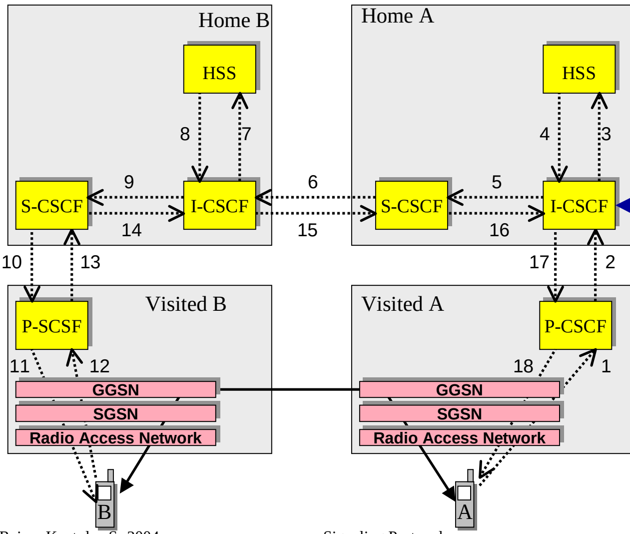


Different Kinds of CSCFs



Proxy CSCF:
Provides
-emergency service breakout,
- triggers for locally-provided services, and
- number normalizing (per local dialing plan)
- Policy Decision point

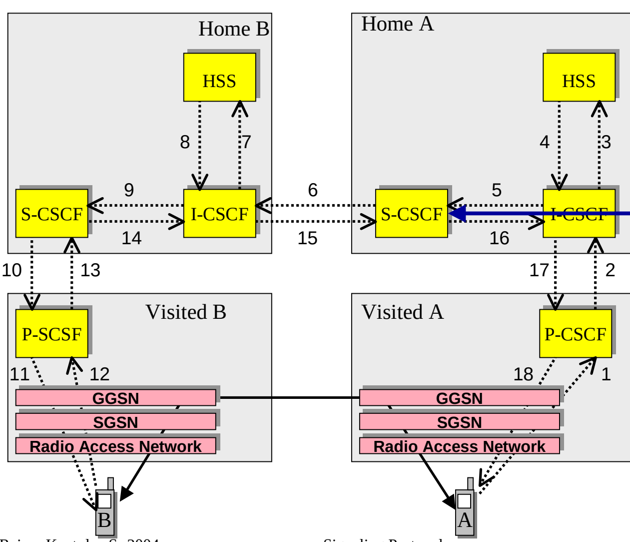
Different Kinds of CSCFs



Interrogating CSCF: Queries the HSS to find the correct S-CSCF. First point of contact for incoming call signalling.

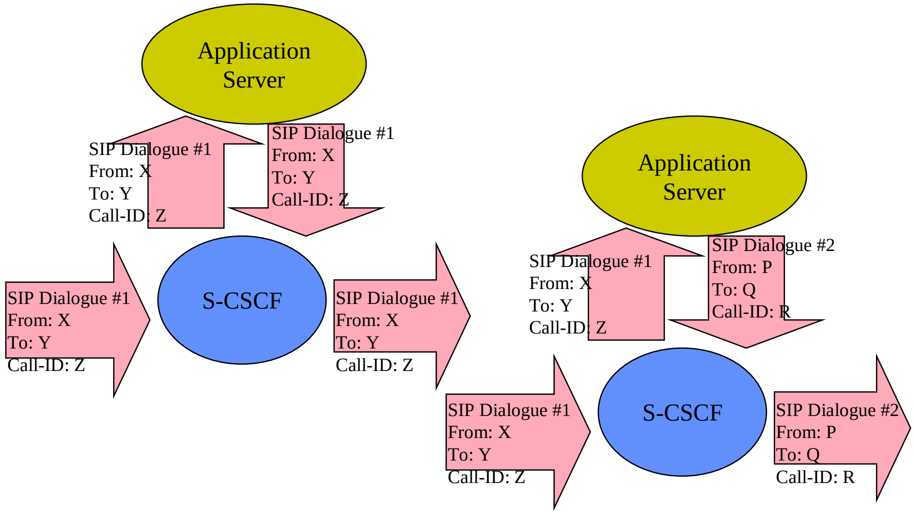
Load distribution node!

Different Kinds of CSCFs

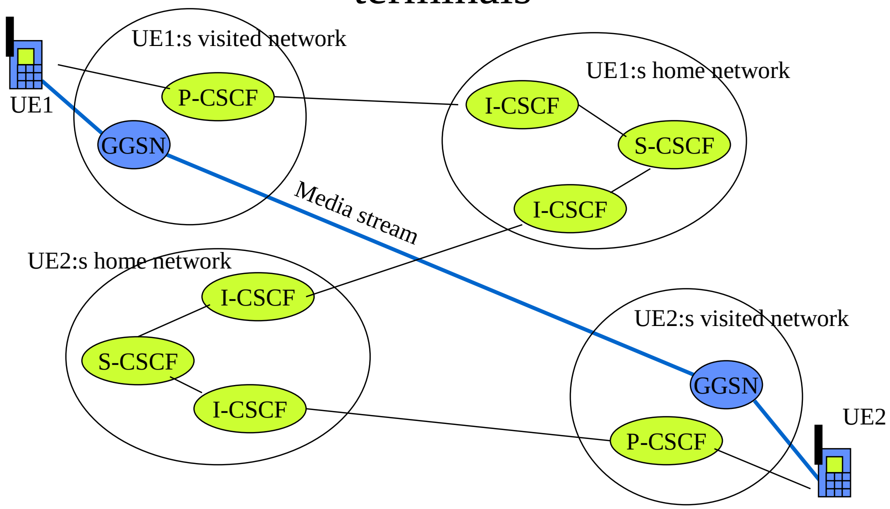


Serving CSCF: Provides subscriber services.

SIP Proxy vs B2BUA



Overview of routing between two mobile terminals



How to Program Services

- Call Processing Language
- SIP CGI
- SIP Servlets
- SIP JAIN (JSLEE – Jain Serv Logic Exec Env)
- Soft SSF and INAP/CAP
- Parlay
- OSA

=> Whatever… Different abstraction levels

==>
There will be many competing ways to implement services!

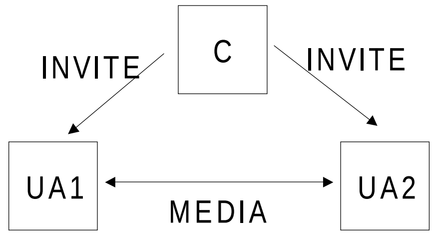
The claim is that it should be as open as flexible as creating services in the web these days

Server types for different services

- Media Server (SIP, RTSP, HTTP)
  – Announcements, IVR, Voicemail, Media on demand
- Conferencing Server (SIP)
  – Media mixer
- Presence Server (SIP)
  – Users status info, capabilities, willingness to communicate
- Web Server (HTTP), E-mail Server (SMTP), Messaging Server (SIP?), Text-to-Speech Server etc.
- Controller Server
  – Co-ordinates the overall service

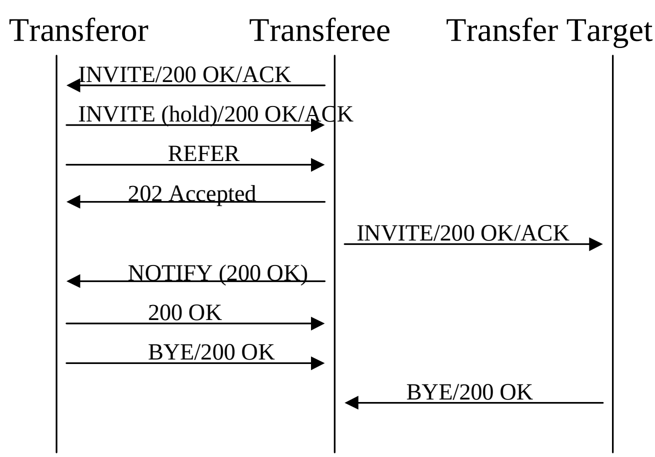
=> Server resources can be addressed by URLs, no need for tight coupling a la MGCP/Megaco

Third Party Call Control is based on SIP



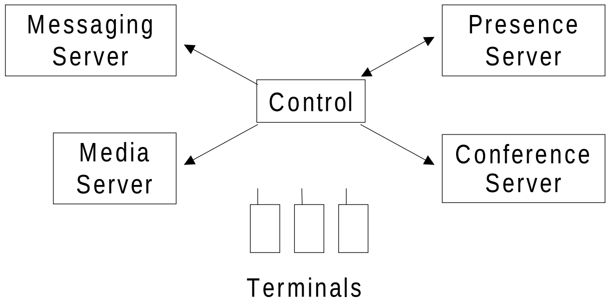
- Details are still to be solved in the IETF
- Powerful tool e.g. for inviting users to centralized conferences or sessions with a Media Server
- In principle third party call control that has never been properly implemented in CSN, is as natural in SIP and first party call control because SIP is used also on the the interface to Applica servers.

REFER and Call Transfer



Media can always go directly from Transferee to Transfer target.

Auto-conferencing Service Example



1. One user orders the conference by filling a web form
2. Controller subscribes to each participants presence
3. When all available, send message or start IVR session to each participant to confirm willingness
4. Connect each participant to conference server. Play announcements to conference from media server when new parties join

Technical Problems

- How to make "service routing"?
- How to make service components really independent?
- If there is dependency, how to move parameters between the components?
- How to secure call release?

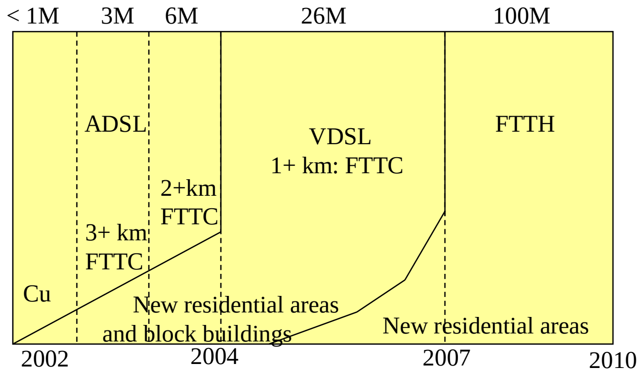
Business problems

- Broadband + VOIP will kill PSTN, this is painful for Incumbent Operators. There is no incentive to deploy VOIP aggressively.
- At the same time voice is becoming mobile.
  - e.g with very conservative mobile policy, ca 90% of call costs are incurred by mobile services in Universities and Politechnics in Finland.
  - Many people have little faith in any wireline voice service.
- How to retain control over Subscribers that have BB connection. Any third party can provide VOIP (with QoS problems not solved).
- Why would Mobile Operators deploy IMS and SIP for voice services when the CS subsystems provides all the needed voice services?
  - it may be that IMS will first be used for services other than VOIP.

Voting for VOIP

- Vendors have stopped developing CS telephony.
- BB deployment is proceeding: Examples of South-Korea, US.
- With wide spread BB, if operators do not deploy VOIP, someone will.

Broadband in South-Korea

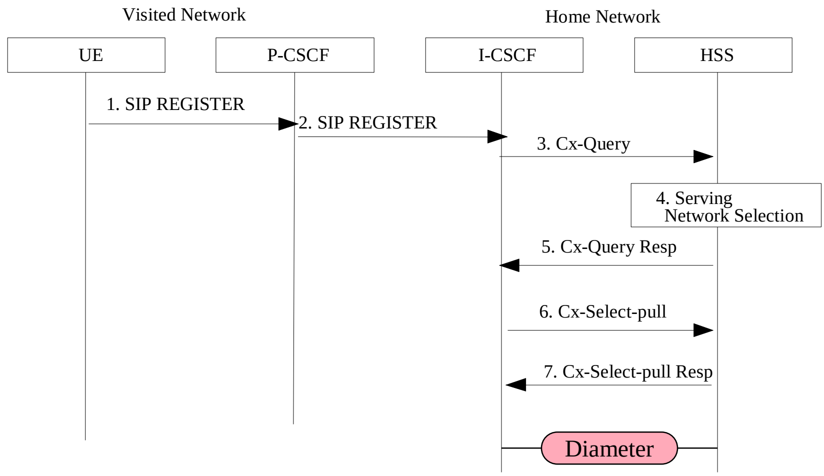


Stats: 12/2002:
- > 70% homes have BB
- > 10M xDSL + CATV+ homes, containing 6M xDSL + 3,7M CATV 0,8 M VDSL
- 10% retail business in Internet
- 50% subscribers < 2km loop 80% < 3 km, avg = 2,5km.

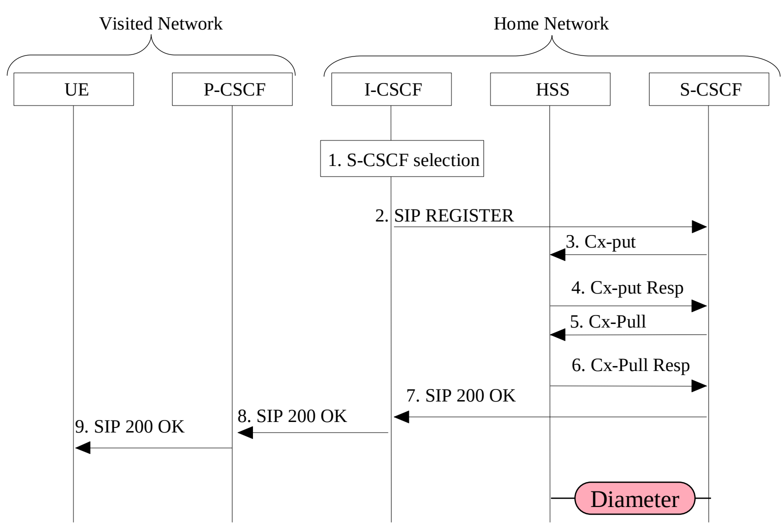
Source: Korea Telecom authors in IEEE Communications Magazine, Dec 2003

Appendix B – 3GPP IMS call flows

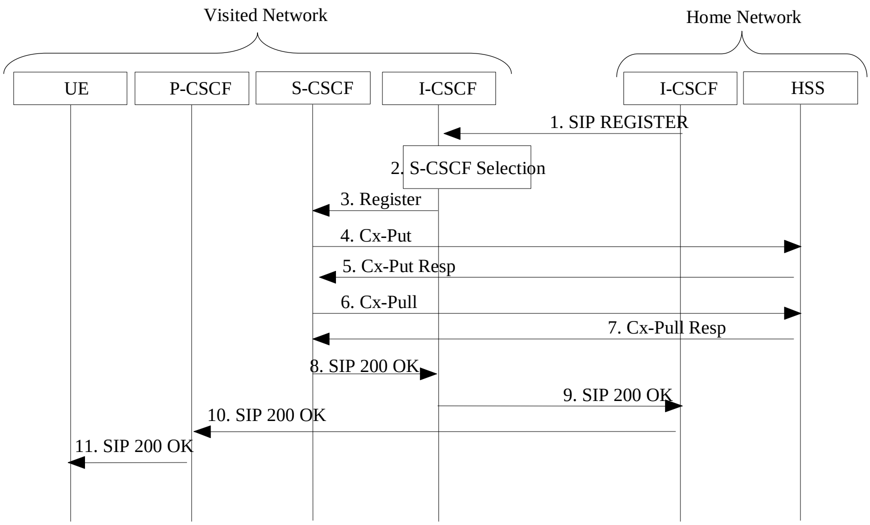
IP Multimedia Registration 1.



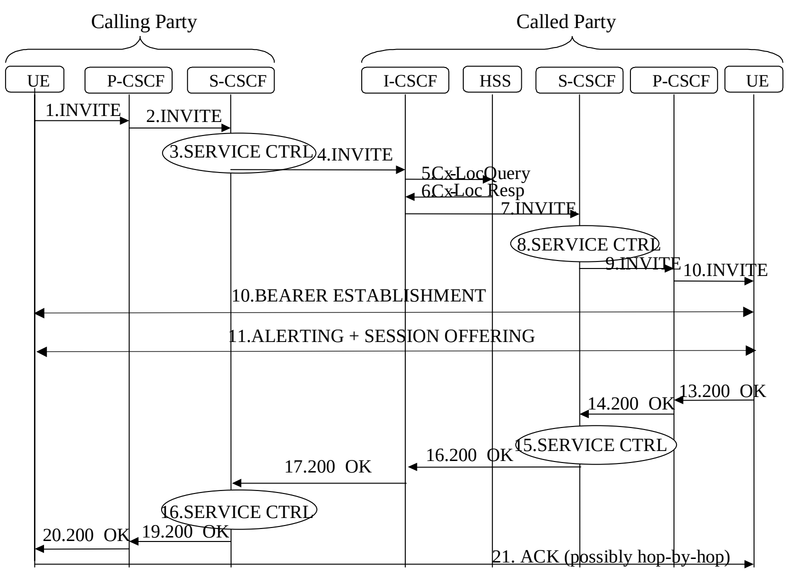
IMS Registration 1a. - S-CSCF in home network



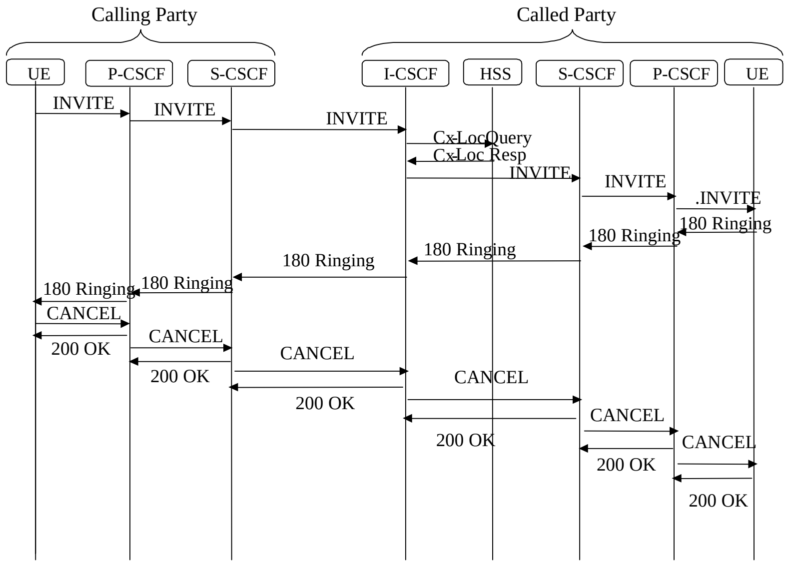
IMS Registration 1b. - S-CSCF in visited network



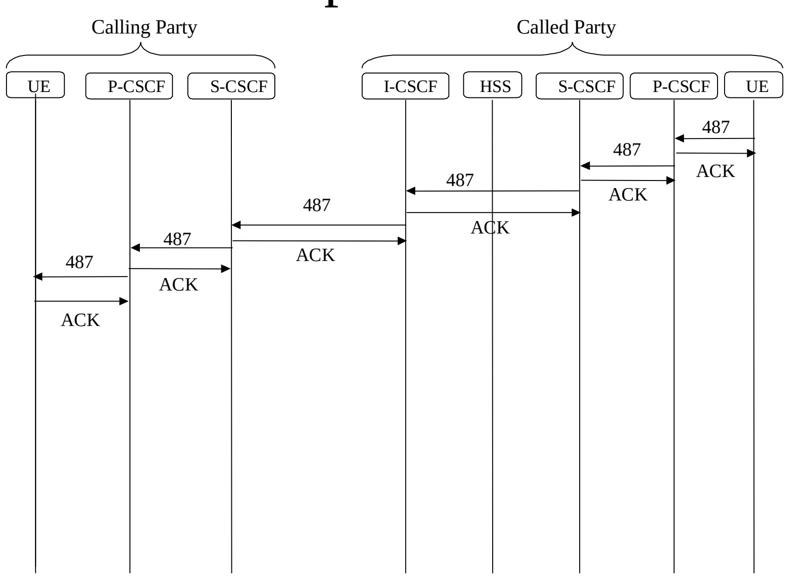
Mobile to Mobile Call



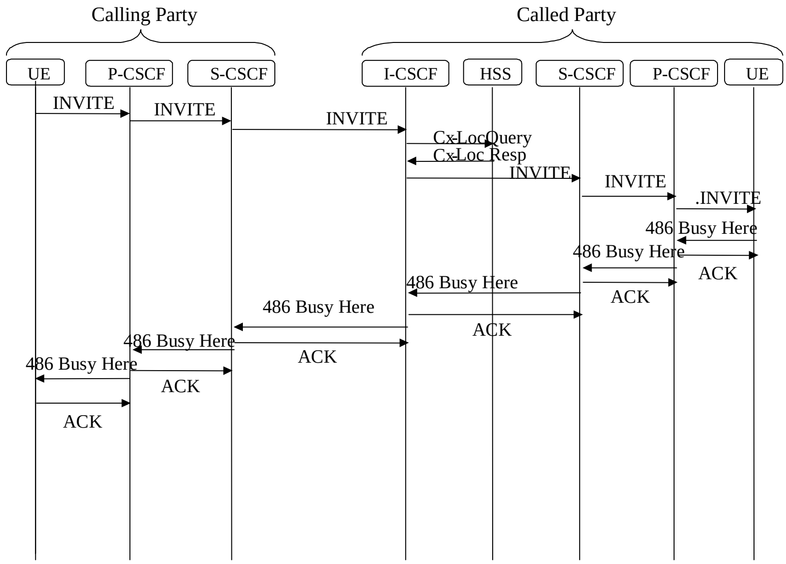
Call flow examples 1. - no answer



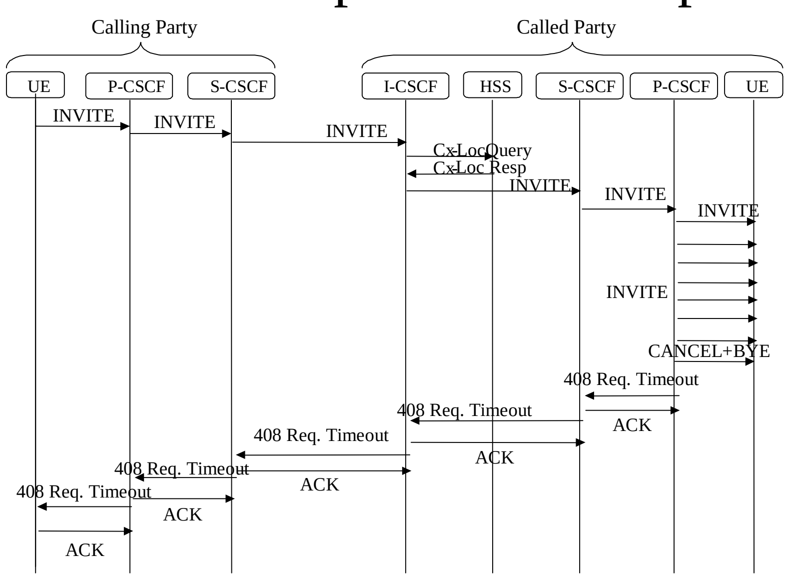
Call flow examples 1. - no answer 2.



Call flow examples 2. - busy



Call flow examples 3. - no response



Call flow examples 4. - temporarily unavailable