SIP Demystified

Book Overview

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- Packet Switching, IETF
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Signalling in CS Network

- CS
  - Strenghts: Fast, Small Latency, QoS guaranteed
  - Weaknesses: Path establishment, occupies resources 100% of time, intelligence in network needed, resillience
- Signalling – to setup, control and tear-down sessions
  - Evolution: DC and AC analog->In-band (FDM transport)->Digital (TDM transport)
    - Access signalling (between terminals and exchange)
      - DTMF
      - Pulse
      - DSS-1 (Digital Subscriber Line No 1) (ISDN, GSM)
    - Trunk Signalling (Between exchanges)
      - CAS (Channel Associated Signalling)
      - CCS (Common Channel Signalling)

CAS and CCS

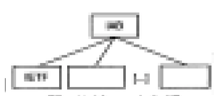
- CAS
    - 16th channel in PCM link
    - Some signalling in voice channels (Caller ID, Callee ID, ...)
- CCS
    - Separate, dedicated signalling network exist
    - One signalling channel handles thousands voice timeslots
    - SS7 dominant CCS signalling
    - Services: toll-free calls, etc
  - SS7
    - Circuit-related
      - E.g. ISUP (Basic Services, circuit management, supplementary services)
      - ISUP has national variations – gateway needed
      - Problem: Intelligence in network, dumb terminals
      - Hard to introduce new services (not flexible architecture)

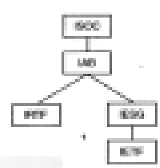
Switching, IP, IETF

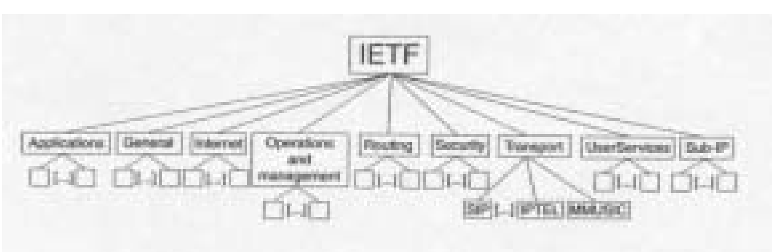
- Packet Switching (efficient, cheaper, more delay, overhead)
  - Datagrams
    - Routing based on destination address
    - No state in network
    - Dynamic routing, load balancing
  - Virtual circuits
    - Virtual circuit established
    - Stateful network
- IP – dominant packet network technology
  - IP layer common
    - Various underlying technologies (ATM, Ethernet, Frame Relay, ...)
    - Various applications above (email, web, VoIP) using common IP layer
  - Intelligence on the edge of the network (centrifugism)
    - Entities on edge provide and consume services
    - Stateless and fast core entities
    - End-to-end services, end-to-end security, etc...

IETF

- IETF toolkit
  - bottom-up approach *("one problem – one protocol")*
  - Reusable protocols
  - Protocols are simple, reusable, scalable, rob

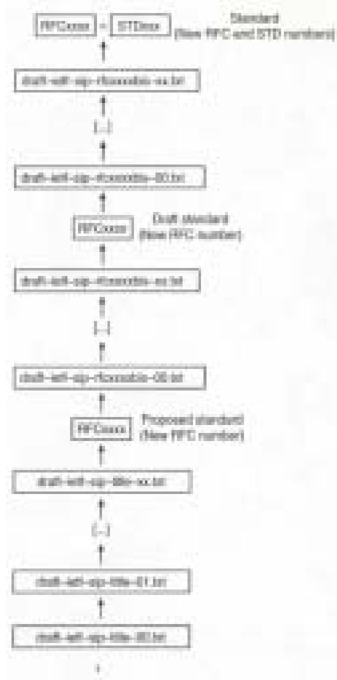






IETF specifications

•Every standard follows the route Proposed standard-> Draft Standard-> Standard



What Protocols Are Needed?

- <u>Signaling protocol</u> to establish presence, locate user, and for session control
- <u>Media Transport Protocols</u> for transmission of media over IP
- <u>Supporting Protocols</u>
  - Gateway location, QoS, AAA,etc…

Multicast

- Several parties involved
  - IPv4 Multicast from 224.0.0.0 – 239.255.255.255
- Saves bandwidth
- Entity that is sending does not have to know all the participants
- Multicast Routing protocols
  - Dense Mode (shortest-path tree per sender)
  - Sparse Mode (shared tree used by all sources)
- IGMP (Internet Group Management Protocol)
  - For hosts that want to become part of multicast group
- Mbone – part of Internet that supports multicast
- RTP – transport of real-time data
  - Sequence number, timestamps
- RTCP – controls RTP transport (every RTP session has paralell RTCP ses.)

QoS – Integrated Serv. and DiffServ

- Integrated Services
  - Different treatment to different flows
  - State info stored in network, routers examine packets!!!(not good)
  - Reservation merging
  - RSVP protocol – for reservation of resources
- DiffServ
  - Defines several traffic classes with different priority levels
  - Packets tagged with level tags at the beginning
  - Routers just examine tags
  - Better scaling

Other Protocols

- SAP (Session Announcement Protocol)
  - Distribute info about multicast sessions on a well-known address and port
- SDP (Session Description Protocol)
  - Describes session, text-based
  - E.g. time, media used, codec used, port used, subject, etc…)
  - Extensible (*a* line gives extension, e.g. *a=volume:8*)
- SDPng (next generation)
- RTSP (Real-Time Streaming Protocol)
  - Simmilar to VCR controls (stop, pause,play, record)
- RSVP – ReSerVation Protocol
  - For resource reservation in the network

Session Initiation Protocol - SIP

- Originally designed to invite users to Mbone sessions
- In IETF:
  - SIP WG – for SIP specifications and extensions
  - SIPPING WG – for applications that use SIP
- Was RFC2543, now RFC 3261!
- For <u>session establishment</u>, <u>modification</u> and <u>termination</u>
- Independent of media session and on mechanism for describing session
- Used to distribute SDP among potential participants
- Reusable addresses: SIP addresses similar to email addresses
  - E.g. sip:someone@somewhere.com

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
    - Group addresses
  - 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 not defined by SIP specs (e.g. LDAP)

SIP Features

- Part of IETF toolkit
  - Reusing other protocols & mechanisms
  - 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 reusage
- Same routing as SMTP
- Reuses infrastructure (all applications will use SIP entities for different services)

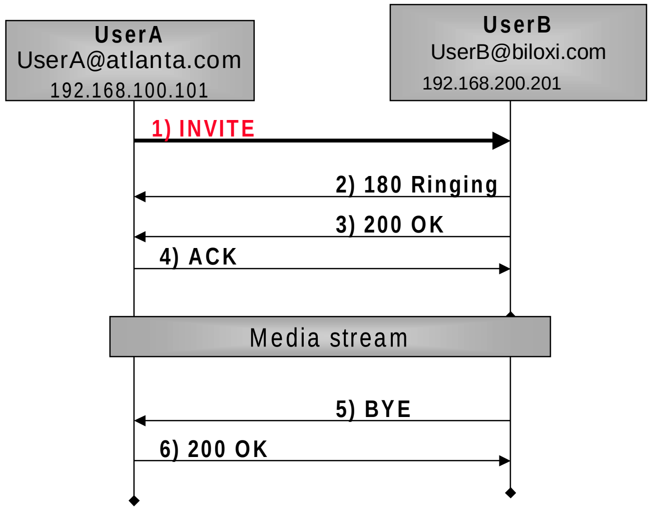
Addressing

- **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… is easy implemented

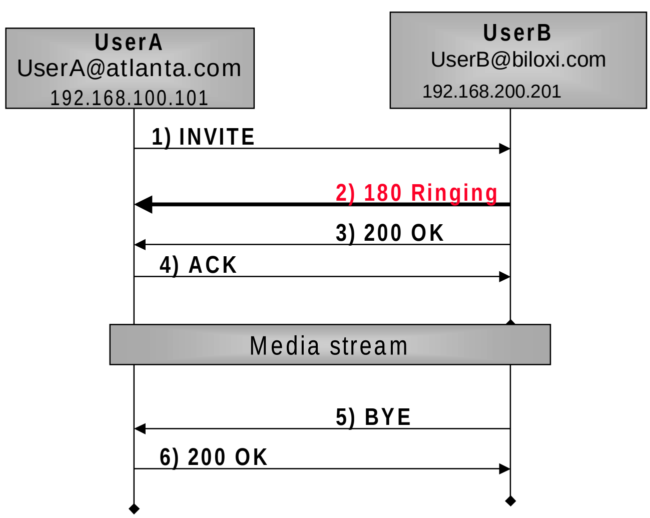
”Basic call” Example

- 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 !

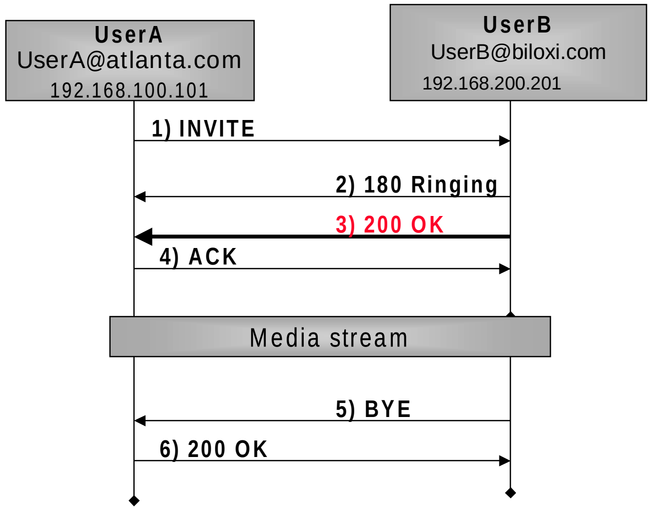
"Basic Call" call flow



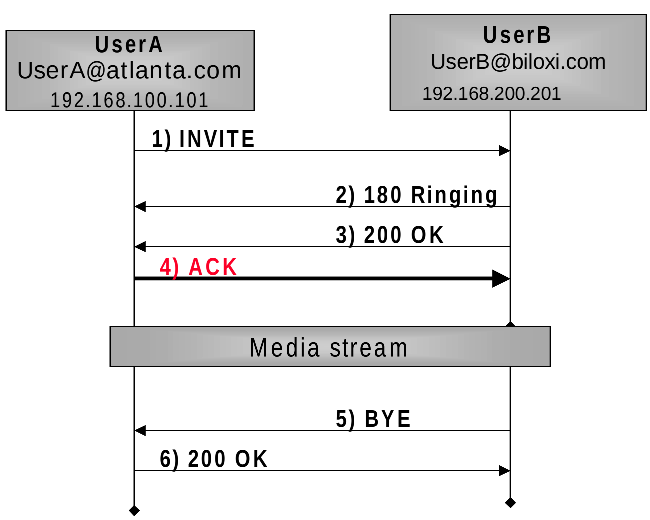
"Basic Call" call flow



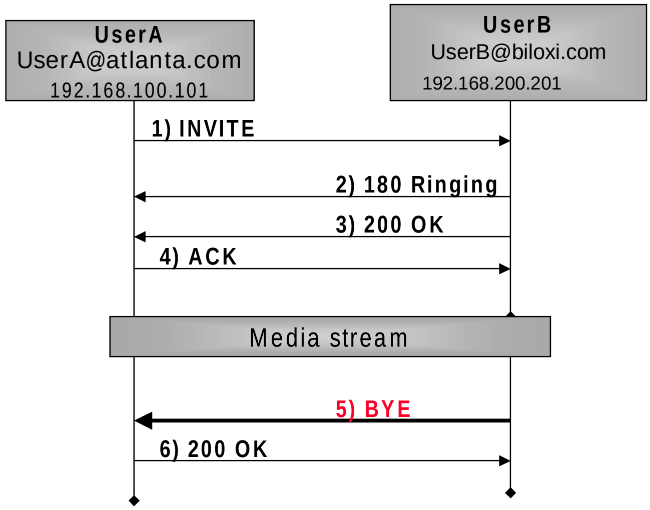
"Basic Call" call flow



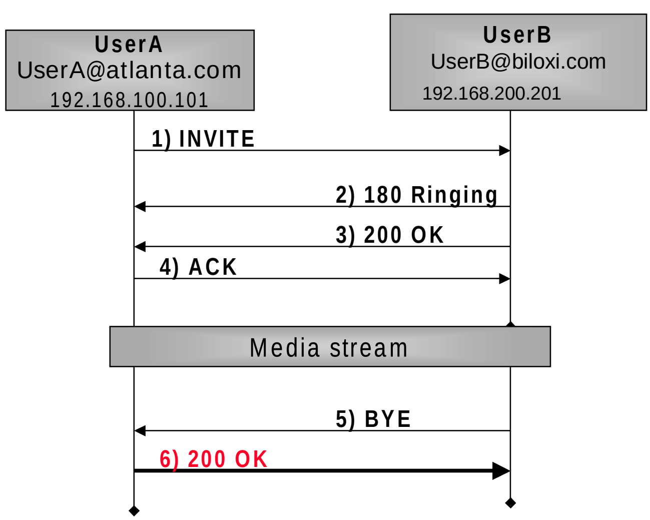
"Basic Call" call flow



"Basic Call" call flow



"Basic Call" call flow



SIP methods (requests)

- SIP methods are invoked on servers when requests arrive:
  - REGISTER requests 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

- 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

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

- 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 come, with all Record-route headers in it (then UAC knows which relays want to stay in signalling**
- **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)

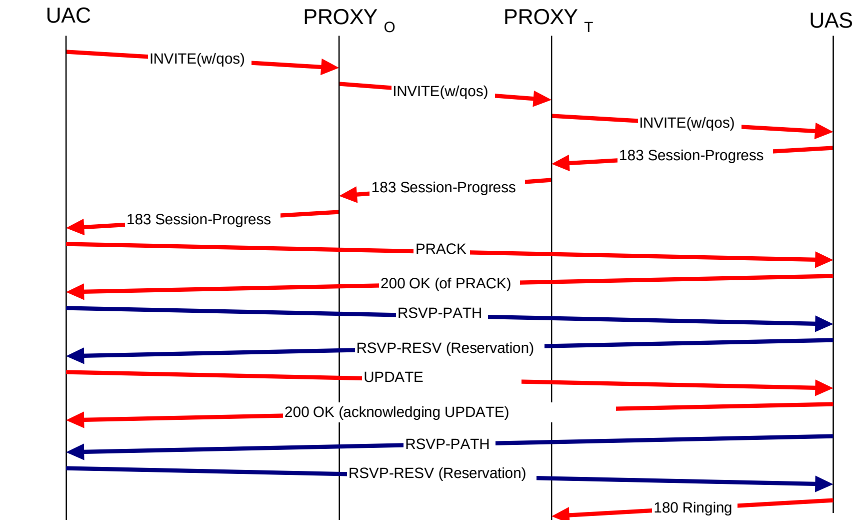
Feature Negotiation (OPTIONS)

- *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

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

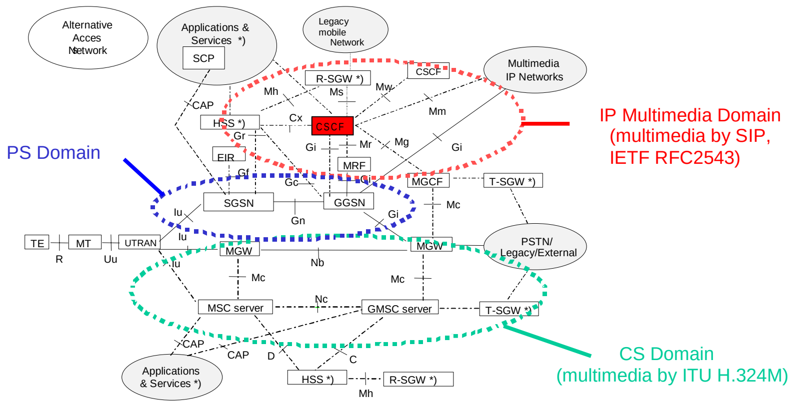
Sample Invite Sequence, with Resource Reservation



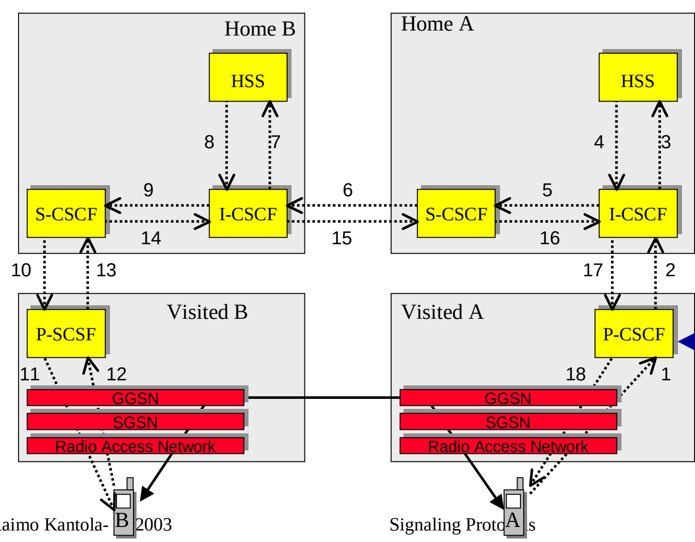
More SIP extensions

- MESSAGE
  - For instant messaging
- INFO
  - To transport mid-session information (very useful in SIP-PSTN gateways)
- Automatic configuration
  - DHCP or Service Location Protocol (SLP)
- Caller Preferences
  - New headers: Accept-Contact, Reject-Contact, Request-Disposition
- SUBSCRIBE/NOTIFY
  - Two new methods for async. notifications. Used in Presence concept
- REFER
  - For session transfer (Refer-To: and Reffered-By:)
- DO (for sending commands to appliances)

3GPP Network Model

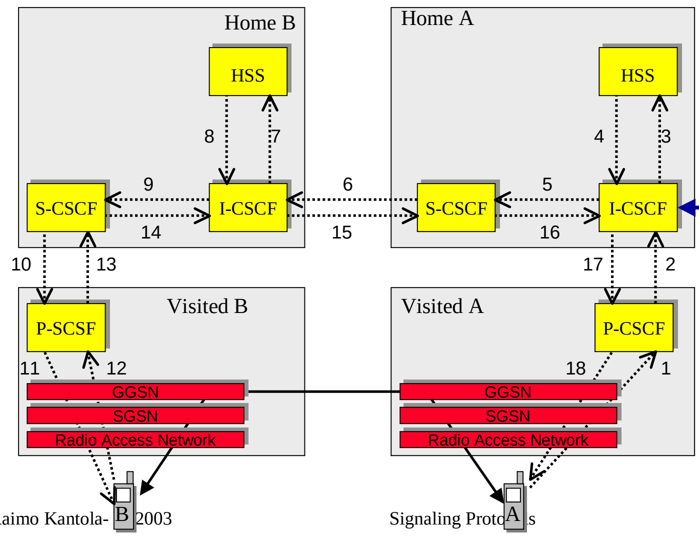


Different Kinds of CSCFs



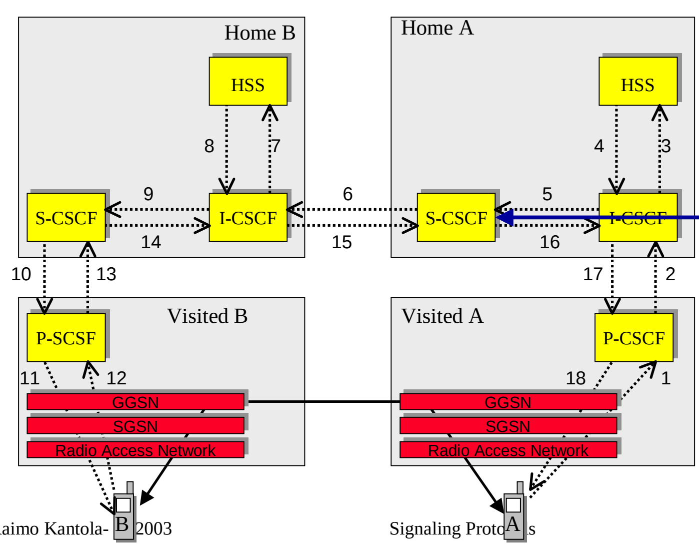
Proxy CSCF: Provides emergency service breakout, triggers for locally-provided services, and number normalizing (per local dialing plan)

Different Kinds of CSCFs



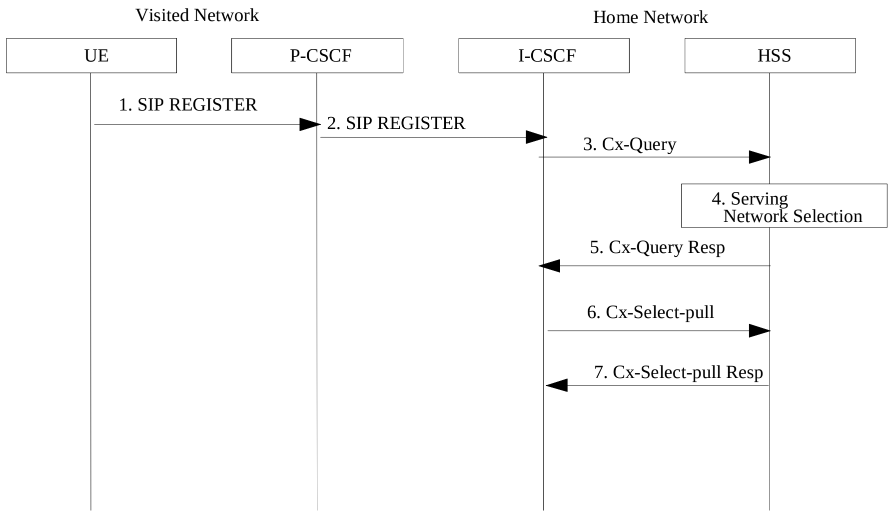
Interrogating CSCF: Queries the HSS to find the correct S-CSCF. First point of contact for incoming call signalling.

Different Kinds of CSCFs



Serving CSCF: Provides subscriber services.

IP Multimedia Registration 1.



SIP and new service architectures

- **Call Processing Language**
  - XML based language to define call service
  - CPL definition is now simple and can be used to describe basic services
- **SIP/CGI**
  - SIP Application Server could implement a CGI interface to describe services
- **SIP Servlet API**
  - SIP Servlet Application Server is comparable in principal to CGI
  - Advantage: use of the java tools and security, more performant than CGI

SIP and new service architectures 2.

- **Java Enhanced SIP (JES)**
  - Defines a way to transport Java applets, objects or agents in a SIP message
  - It could be used to send service logic to a SIP client
  - The Java entity can be embedded in the message if small enough or referenced for later downloading using HTTP or SOAP
- **SIP and WEB/WAP Integration**
  - SIP is working well in conjunction with Web (MIME, URI, DNS). Lot of SIP services could be originated by Web pages (click to call, Web phone book…)
  - SIP and HTTP stacks have a lot in common. Advanced SIP implementation usually provides simple HTTP stack

SIP and new service architectures 3.

- **SIP and WEB/WAP Integration cont.**
  - SIP and other IP services will probably come to 3G also and they will replace WAP services if WAP does not evolve
  - WML could be carried inside SIP messages. WML has effective syntax for menus and SIP is the one and only transport which also include intelligent features.
- **SIP and SOAP**
  - SOAP can be used in conjunction with SIP to offer services. SOAP can be used by a SIP client to discover and access services on the network. SIP application server can use SOAP to access Business Logic and databases in the network.

Appendix A – Call Setup Examples

Registration example with SIP

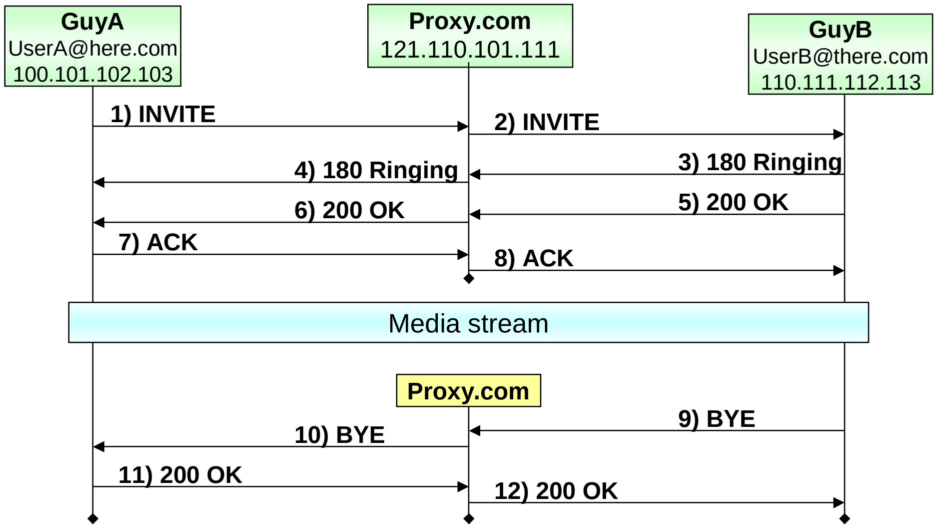
Bob
bob@biloxi.com

biloxi.com

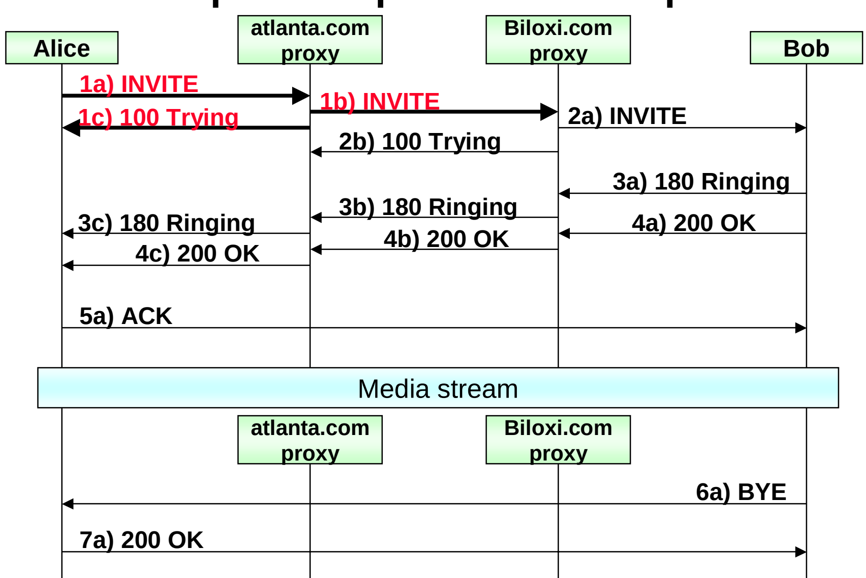
REGISTER sip:registrar.biloxi.com SIP/2.0
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Bob <sip:bob@biloxi.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:bob@192.0.2.4>
Expires: 7200
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7
;received=192.0.2.4
To: Bob <sip:bob@biloxi.com>;tag=2493k59kd
From: Bob <sip:bob@biloxi.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:bob@192.0.2.4>
Expires: 7200
Content-Length: 0

Call Setup example with one proxy



Call Setup example with two proxies



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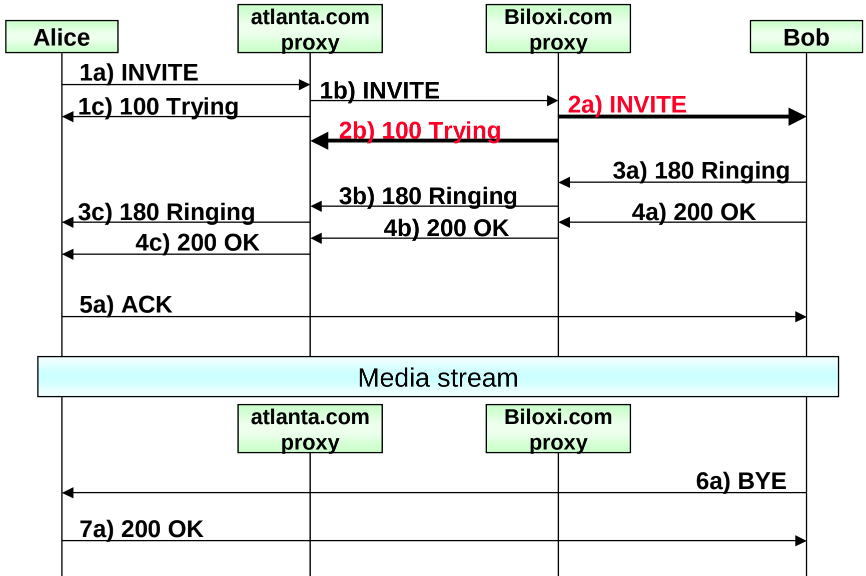

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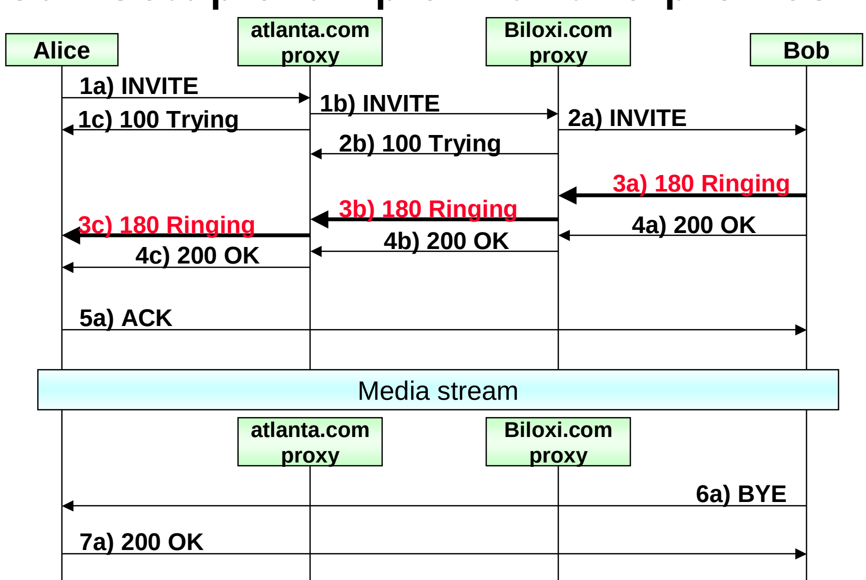

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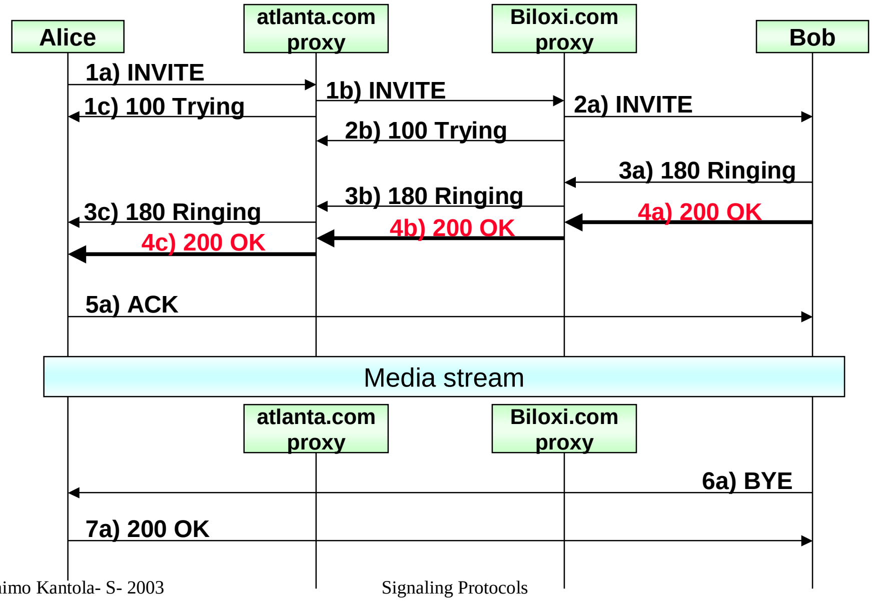

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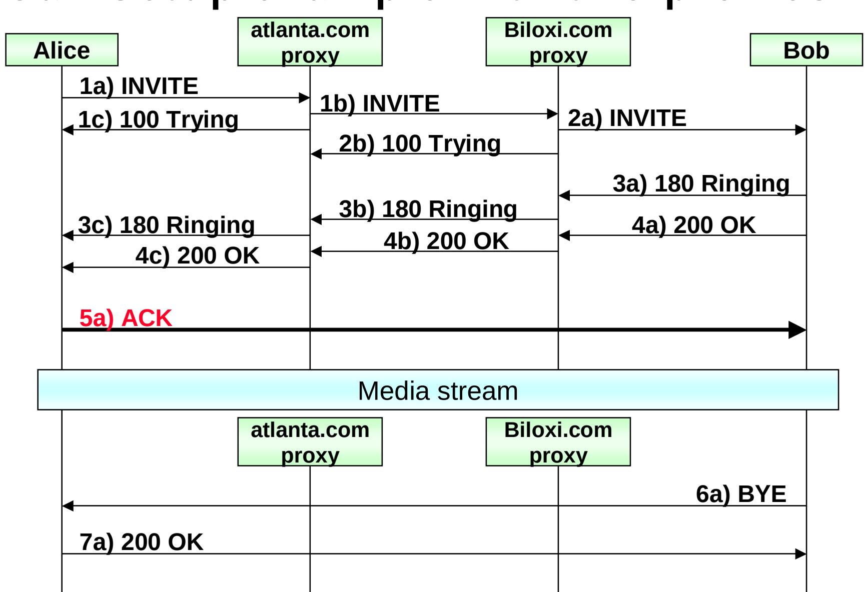
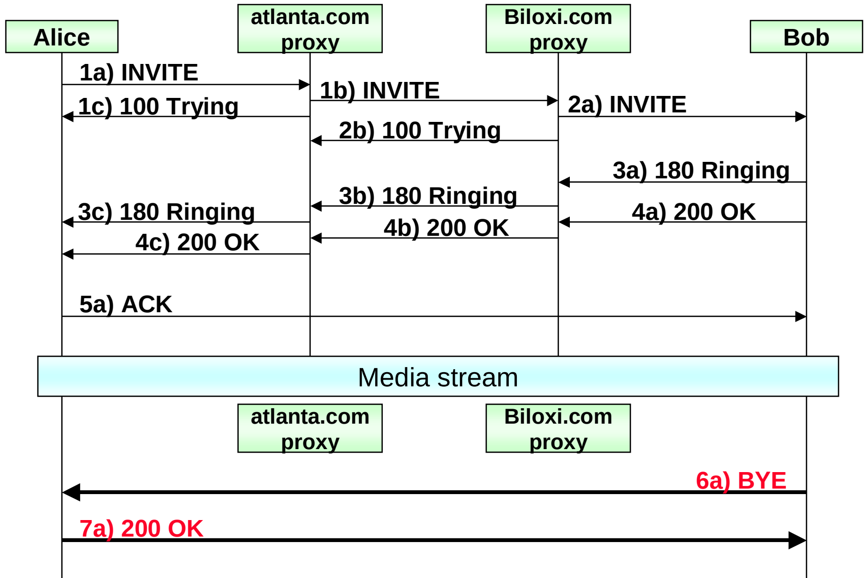
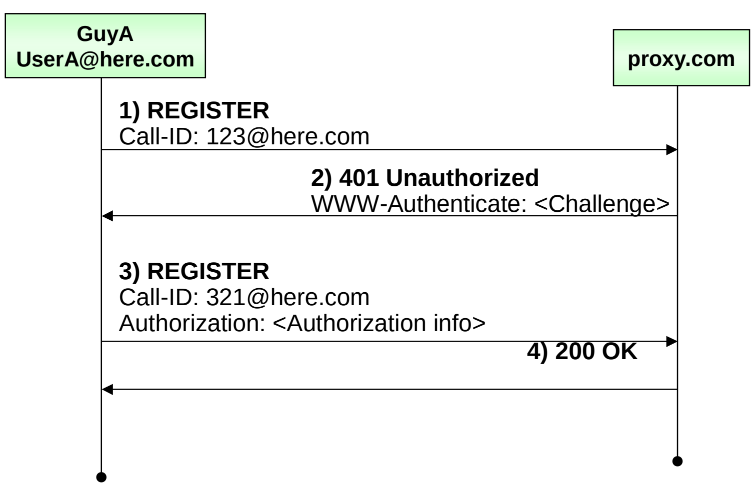


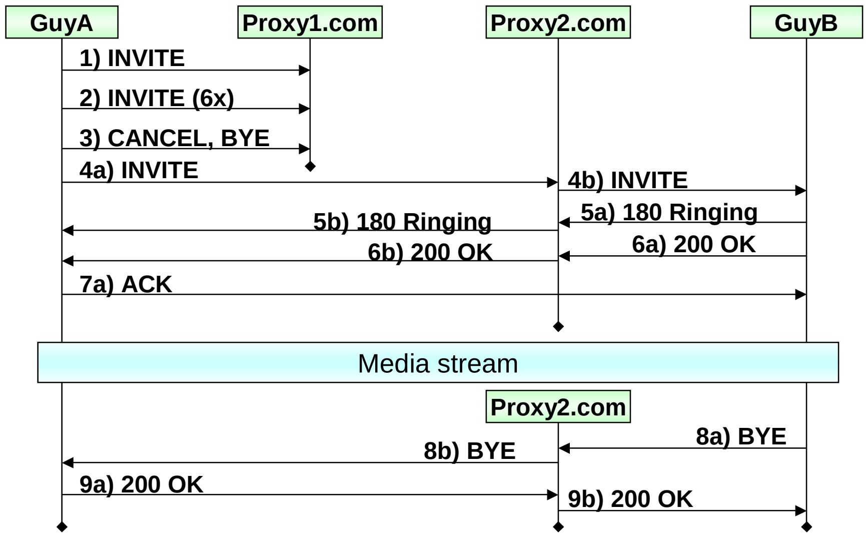
Call Setup example with two proxies



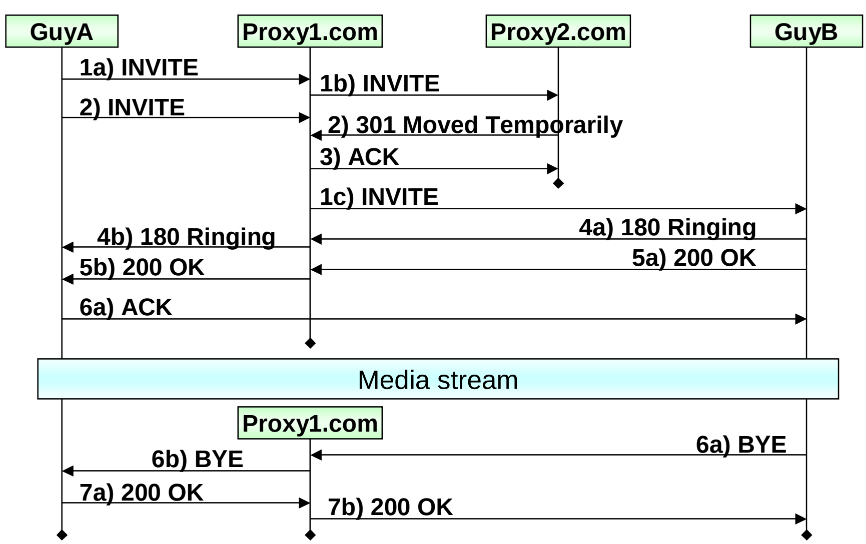
Registration example with SIP authentication



Call Setup example with a non-working proxy

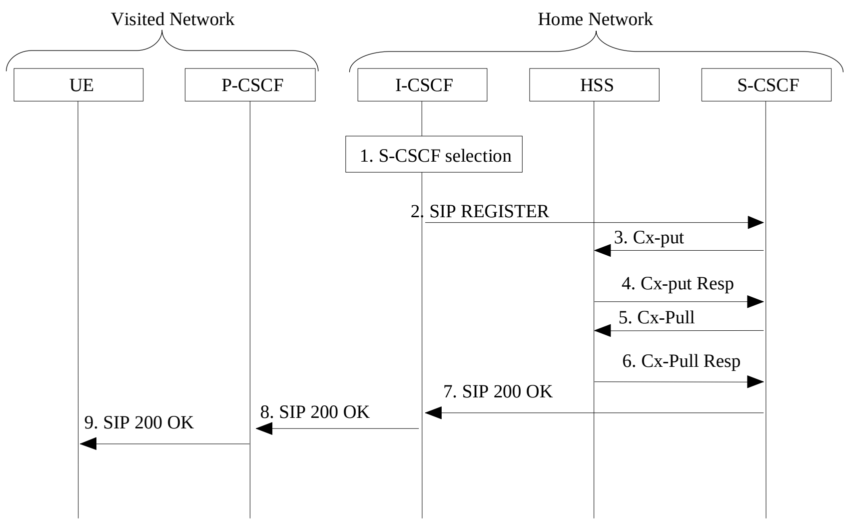


Call Setup example with a Redirect server

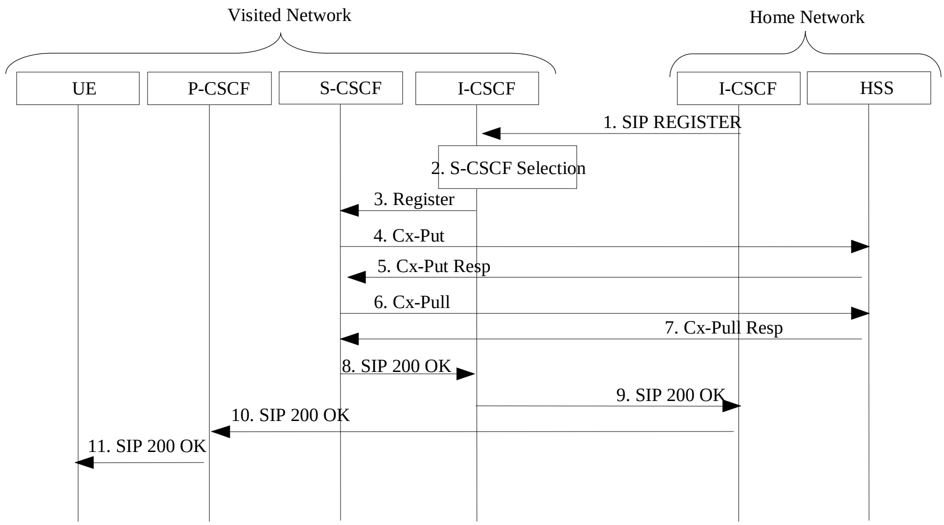


Appendix B – 3GPP IMS call flows

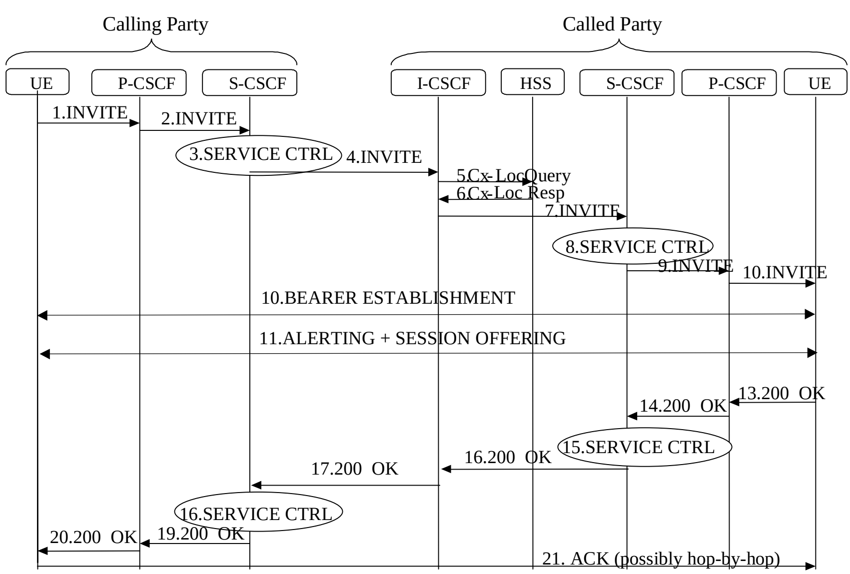
IMS Registration 1a. - S-CSCF in home network



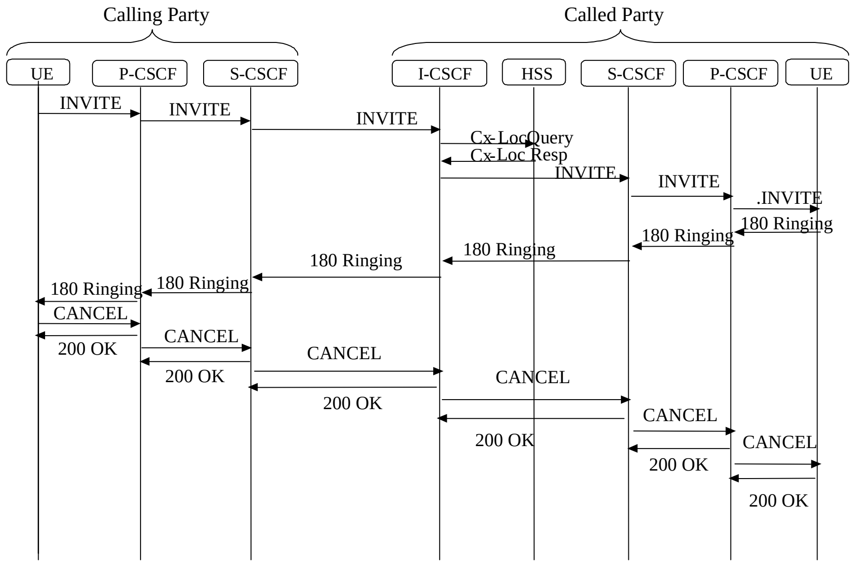
IMS Registration 1b. - S-CSCF in visited network



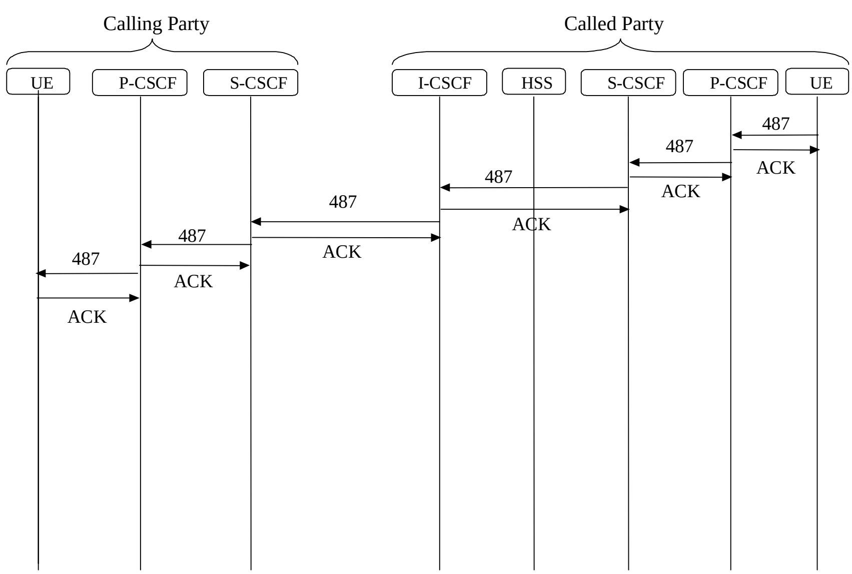
Mobile to Mobile Call



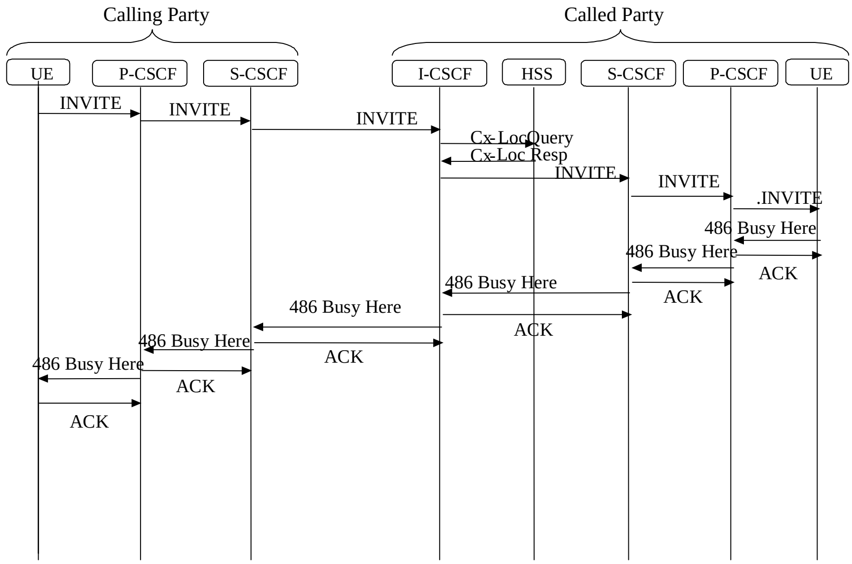
Call flow examples 1. - no answer



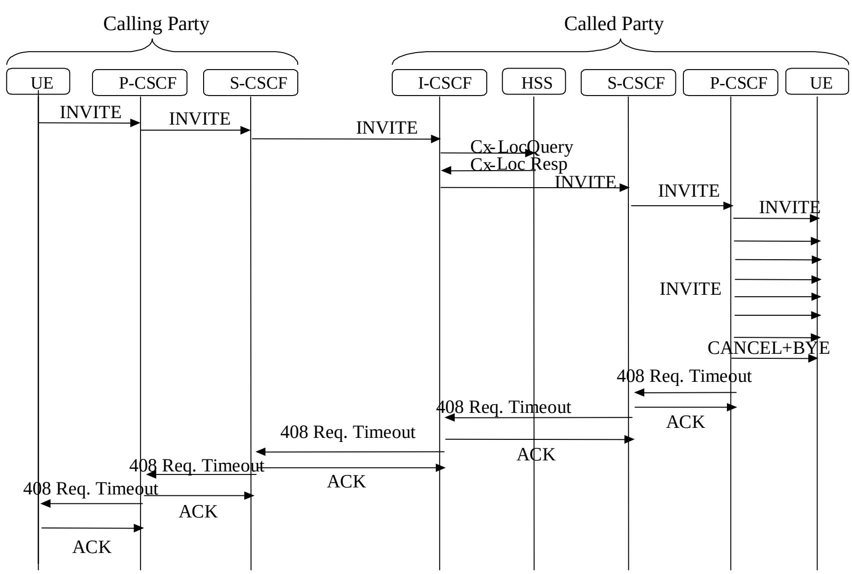
Call flow examples 1. - no answer 2.



Call flow examples 2. - busy



Call flow examples 3. - no response



Call flow examples 4. - temporarily unavailable