Call Setup Examples based on Generic SIP

SIP in 3G

Registration example with SIP

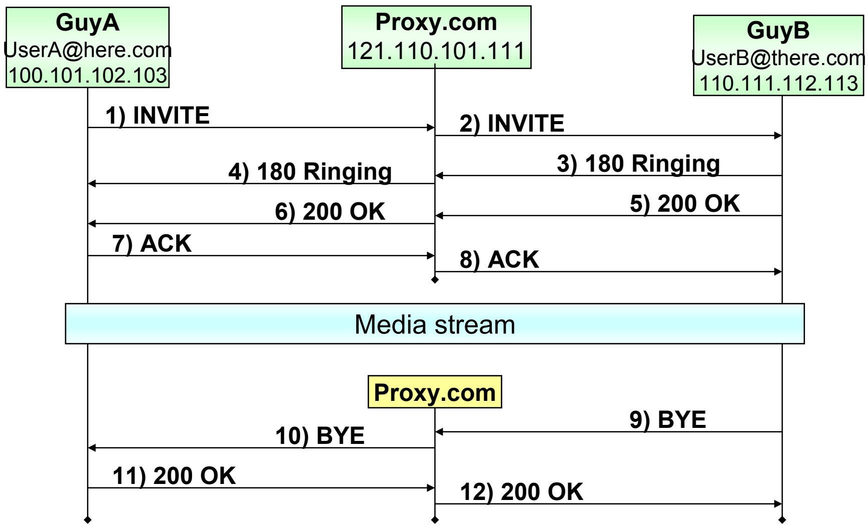
Bob
bob@biloxi.com

biloxi.com

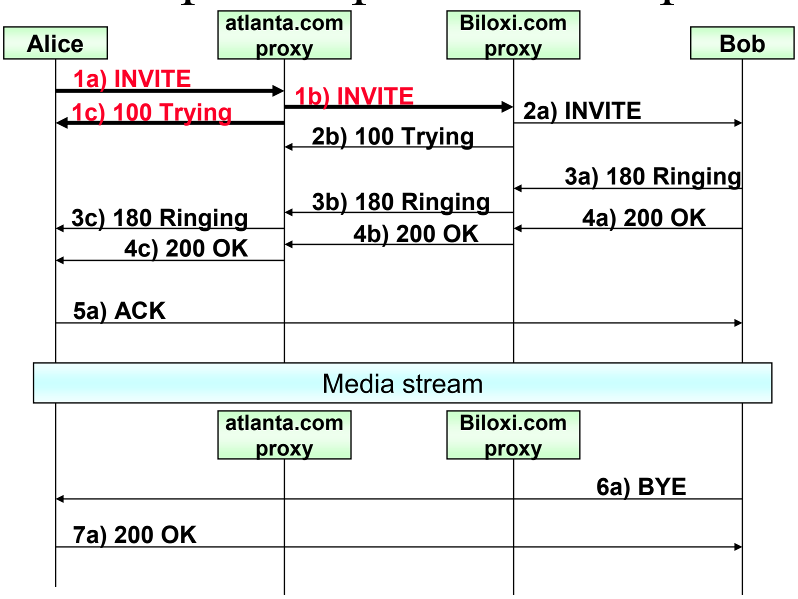
```
REGISTER sip:registrar.biloxi.com SIP/2.0
 Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7
 Max-Forwards: 70
 To: Bob <sip:bob@biloxi.com>
 From: Bob <sip:bob@biloxi.com>;tag=456248
 Call-ID: 843817637684230@998sdasdh09
 CSeq: 1826 REGISTER
 Contact: <sip:bob@192.0.2.4>
 Expires: 7200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

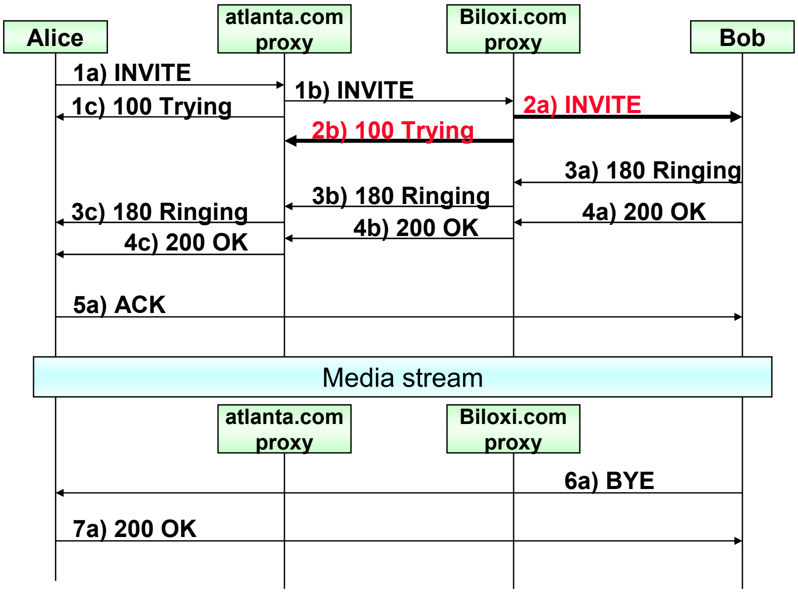
Call Setup example with one proxy



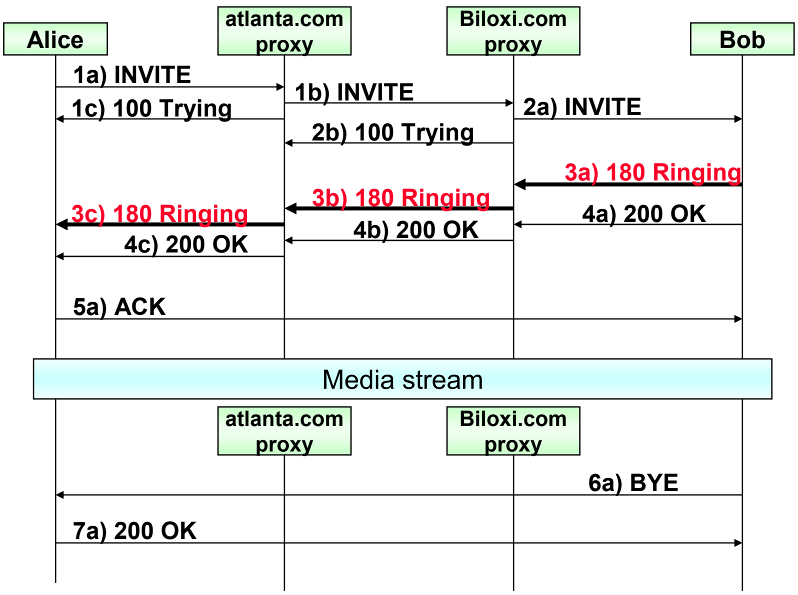
Call Setup example with two proxies



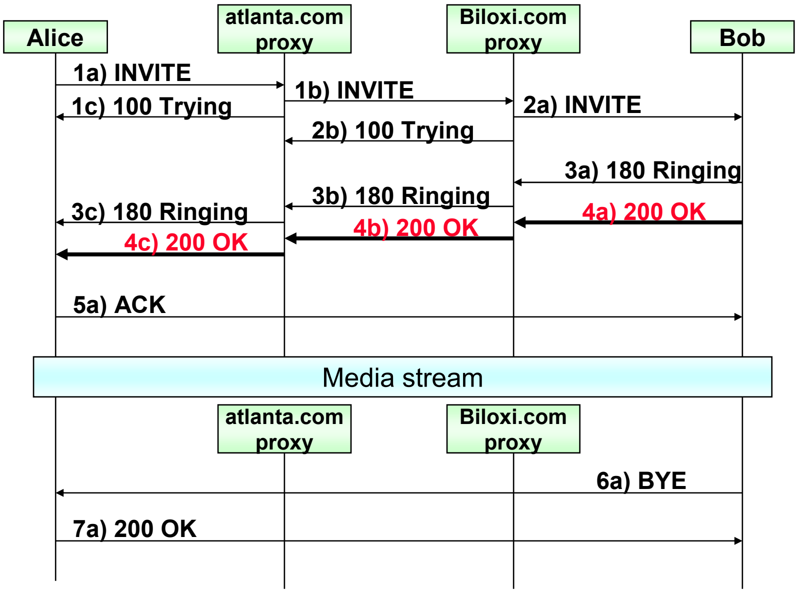
Call Setup example with two proxies



Call Setup example with two proxies



Call Setup example with two proxies



Call Setup example with two proxies

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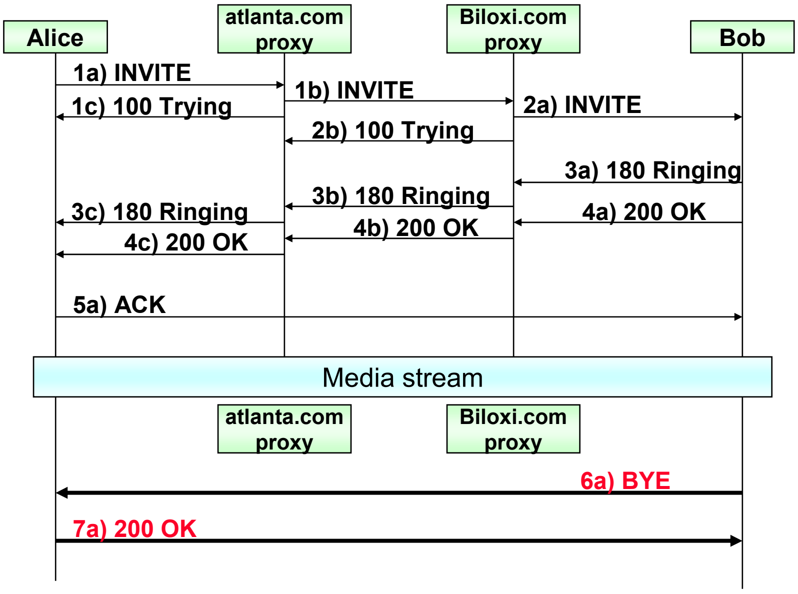
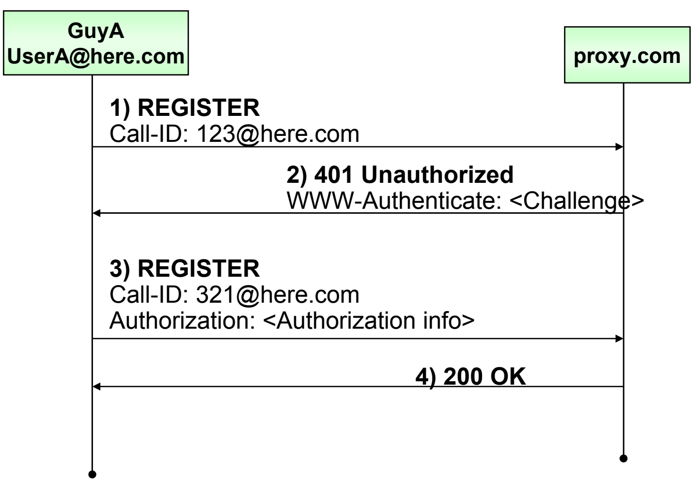
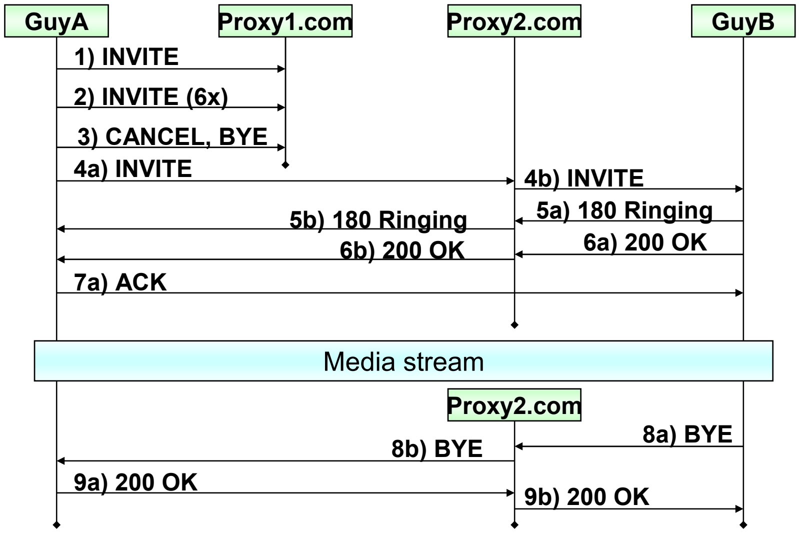


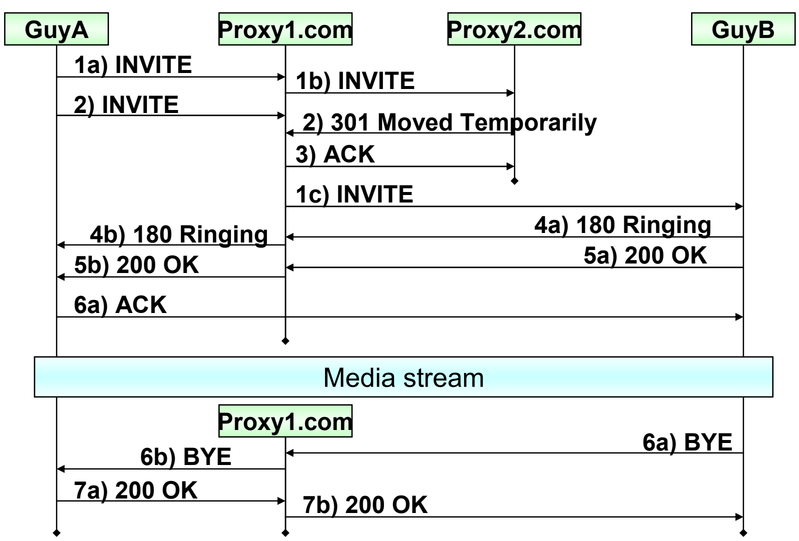
Registration example with SIP authentication



Call Setup example with a non-working proxy



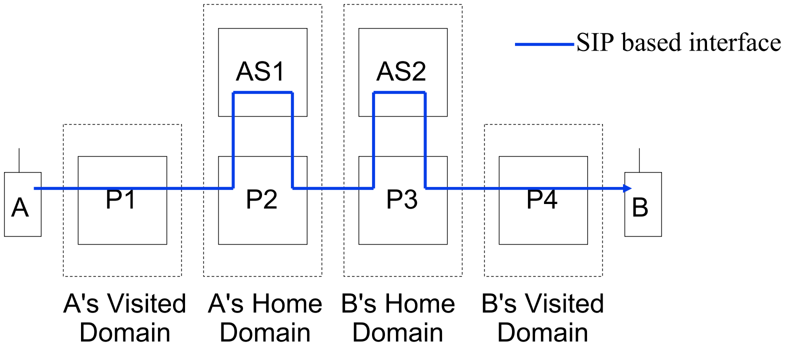
Call Setup example with a Redirect server



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

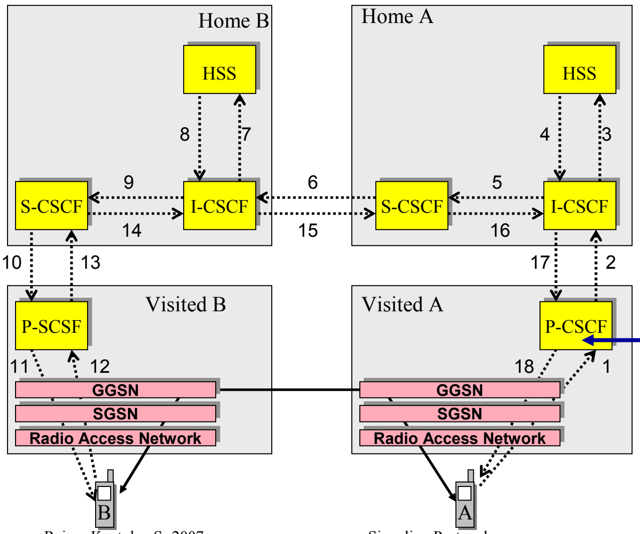
SIP Entities & Service Capabilities

- 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 e.g Sessio Border Controller or 3G SIP

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 that orchestrate things

Different Kinds of CSCFs



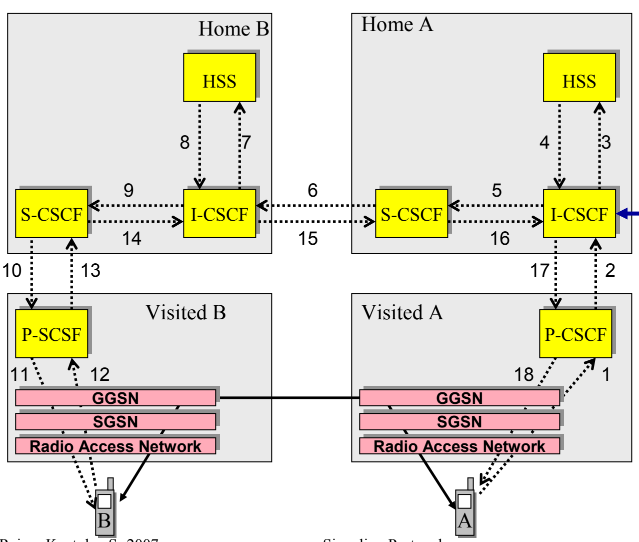
Proxy CSCF:
In the same network as the GGSN: if the visited net does not support IMS, can be in the home network.

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

-user authentication

-maintains a security association with the terminals for signaling

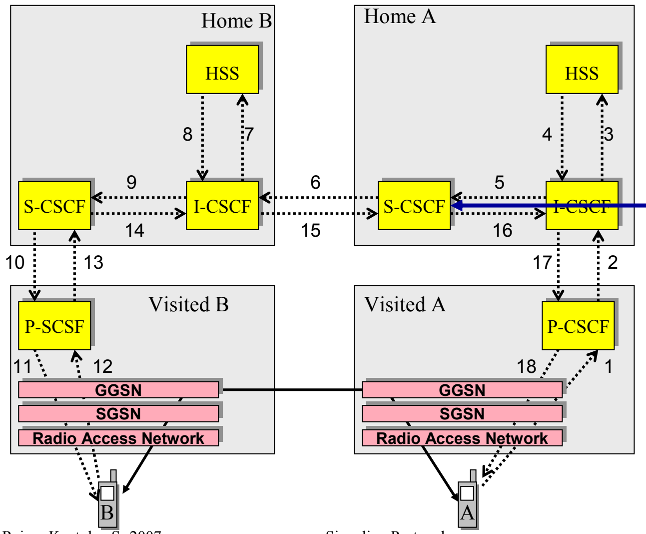
Different Kinds of CSCFs



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

Load distribution node!

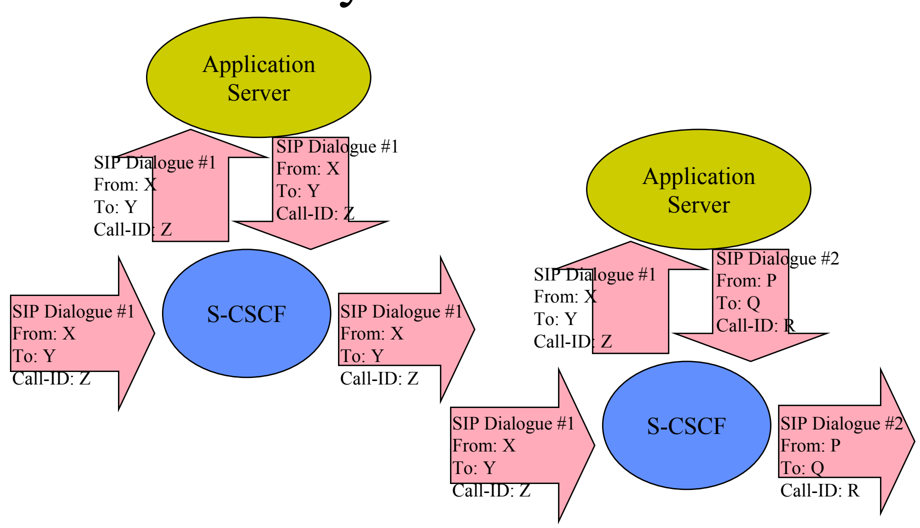
Different Kinds of CSCFs



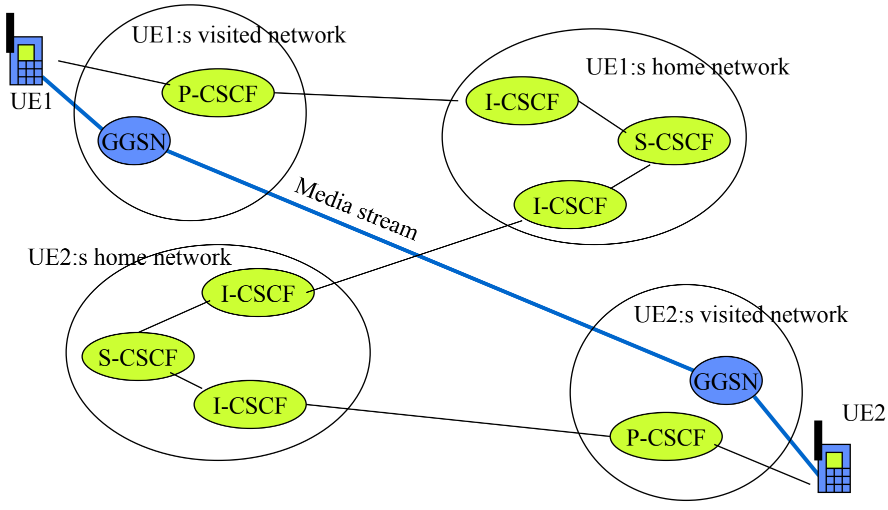
Serving CSCF: Provides subscriber services.

Interface to Application servers.

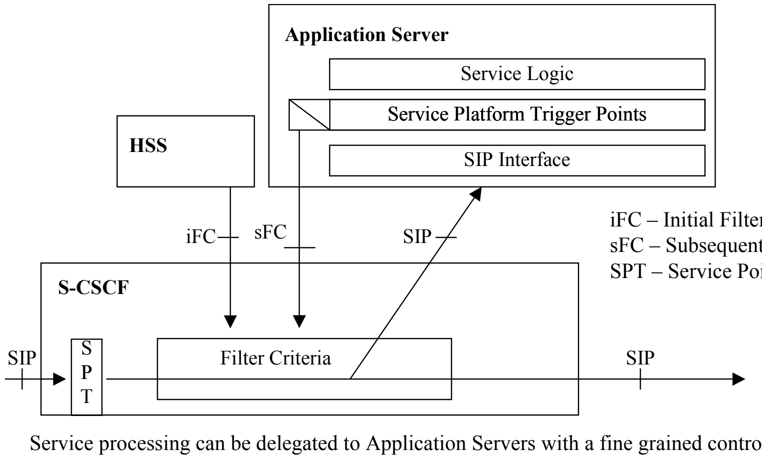
SIP Proxy vs B2BUA



Overview of routing between two mobile terminals



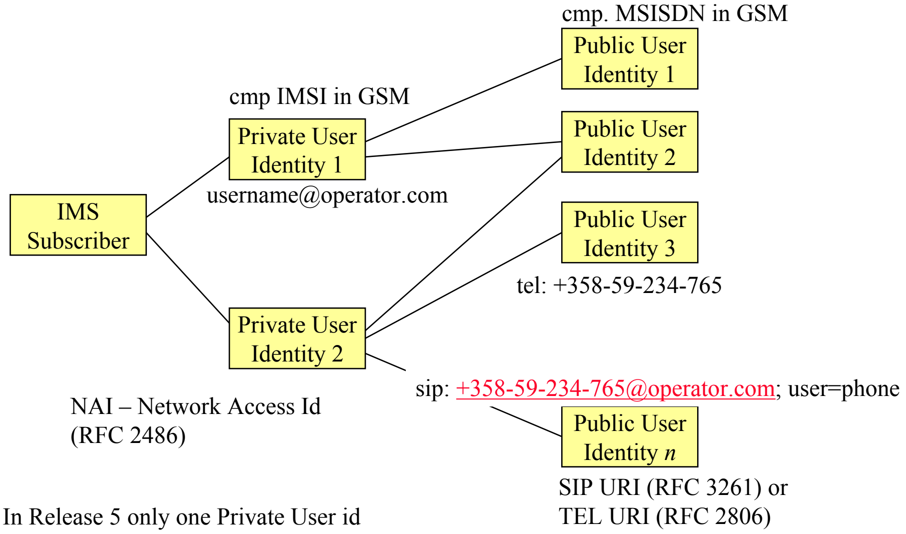
3G Application Triggering



iFC – Initial Filter Criteria
sFC – Subsequent Filter Criteria
SPT – Service Point Trigger

Service processing can be delegated to Application Servers with a fine grained control:
Filter criteria in IMS triggering is bound to user identities, since a user may have many identities, different services may be invoked depending on the identity.

Identification of users in 3G IMS in R6



In Release 5 only one Private User id

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!

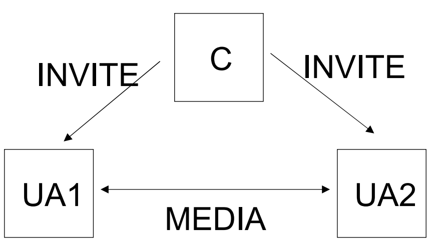
The claim is that it should be as open as flexible as creating services in the web these days

Server types for different services

- Media Server (SIP, RTSP, HTTP)
  - Announcements, IVR, Voicemail, Media on demand
- Conferencing Server (SIP)
  - Media mixer
- Presence Server (SIP)
  - User’s 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

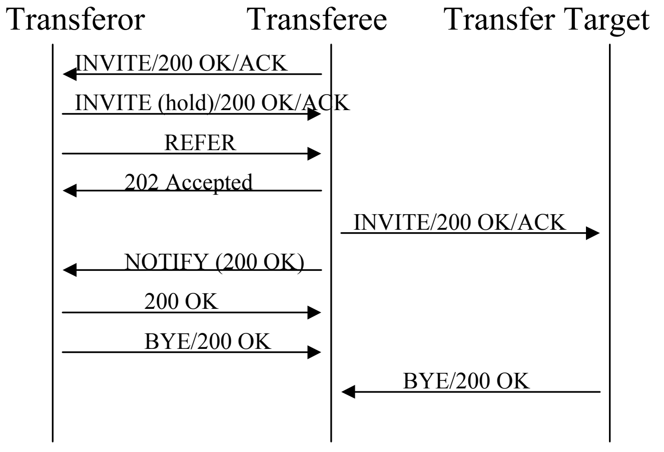
=> Server resources can be addressed by URLs, no need for tight coupling a la MGCP/Megaco

Third Party Call Control is based on SIP



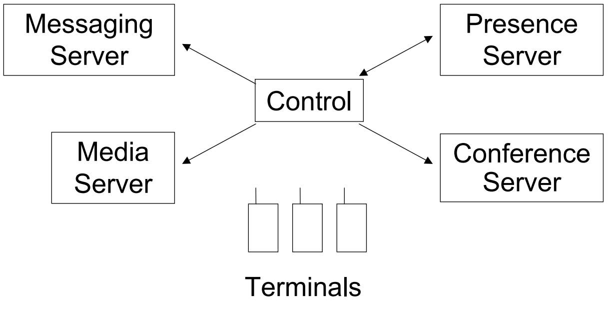
- Powerful tool e.g. for inviting users to centralized conferences or sessions with a Media Server
- In principle third party call control that has never been properly implemented in CSN, is as natural in SIP as first party call control because SIP is used also on the the interface to Application servers.

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 components really independent?
- If there is dependency, how to move parameters between the components?
- How to secure call release?
- Signaling efficiency for narrow band services
  - Problem for narrow band networks and for shared capacity networks when SIP applied to narrow band services
  - SigComp gives some relief with the expense of cpu cycles and memory (most likely less than 1:10 compression)

Emergency calls in IMS

- Requirements
  - different countries have different requirements and different numbers for Emergency calls (Europe 112, USA 911, Japan 119 etc)
  - US: mobile terminal has to be geographically located
  - Europe: the network has to place the call even if there is no SIM card. Call has to be routed to the right Emergency Center.
- IMS issues:
  - GPRS always authenticates the user.
  - Different numbers in different countries → routing problem for roaming customers
- IMS solution in Release 5: The terminal has to place the emergency call using the CS domain in 3G → all voice terminals have to support CS services. P-CSCF has to detect an incoming emergency call by a roaming customer irrespective in which country the customer is roaming and even if the P-CSCF is located in the home network.
- Support for Emergency services has been added to IMS later releases but still all Mobiles support CS services…

Emergency calls in VOIP

- Requirement: The Emergency Center has to see the address of the caller to the emergency number.
- In PSTN the telehone extension has a location number that identifies the copper wire to the residence. The directory number of the caller can always be mapped to the location number and the address of the caller retrieved from a subscriber database.
- IP networks do not support location numbers. IP addresses are allocated to users dynamically. If the user is calling from home, the home address can somehow be identified from a DB. If the user is connected while away from home, VOIP may give a wrong address to the Emergency Center.

Business issues and opportunities

- Broadband + VOIP will kill PSTN, this is painful for Incumbent Operators. There is no incentive to deploy VOIP aggressively.
  - PSTN penetration has drobbed to about 30% in Finland
- 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.
- IMS may be used to provide corporate telephony services – intergration of corporate communication services with other business process IT systems.

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 (e.g. SKYPE).

OECD Broadband Statistics to June 2007

Over the past year, the number of broadband subscribers in the OECD increased 22% from 181 million in June 2006 to 221 million in June 2007.

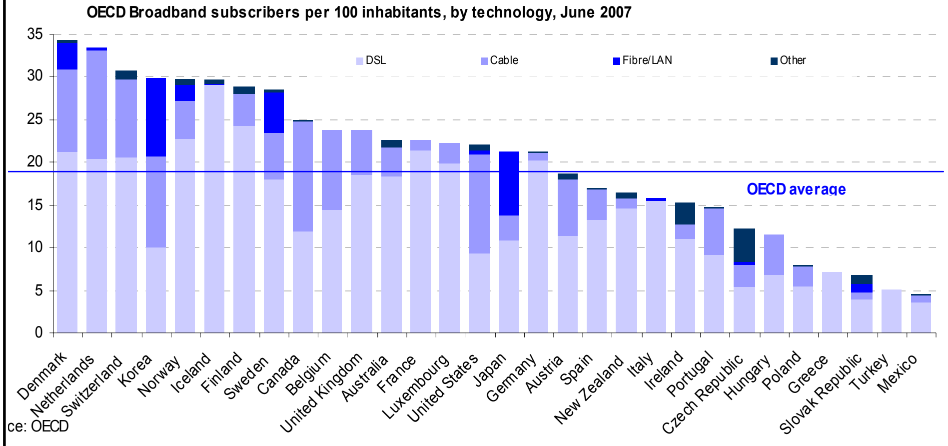
Fibre to the home is becoming increasingly important for broadband access, particularly in countries with high broadband penetration.

DSL:	Cable modem	Fibre	other
62%	28%	8%	1… 2%

Rank	Country	% DSL	% Cable	%Fibre/LAN	oth	Subs/100 inhab	Total subs
1	Denmark	21,3	9,7	2,9	0,4	34,3	1 866 306
2	Netherlands	20,4	12,7	0,4	0,0	33,5	5 470 000
3	Switzerland	20,5	9,3	0,0	0,9	30,7	2 322 577
4	Korea	10,1	10,6	9,2	0,0	29,9	14 441 687
5	Norway	22,7	4,5	1,8	0,7	29,8	1 388 047
6	Iceland	29,0	0,0	0,2	0,6	29,8	90 622
7	Finland	24,4	3,7	0,0	0,8	28,8	1 518 900
8	Sweden	17,9	5,6	4,6	0,4	28,6	2 596 000
9	Canada	11,9	12,9	0,0	0,1	25,0	8 142 320
10	Belgium	14,5	9,2	0,0	0,1	23,8	2 512 884
11	United Kingdom	18,4	5,3	0,0	0,0	23,7	14 361 816
12	Australia	18,3	3,4	0,0	0,9	22,7	4 700 200
13	France	21,4	1,1	0,0	0,0	22,5	14 250 000
15	United States	9,3	11,5	0,6	0,7	22,1	66 213 257
16	Japan	10,8	2,9	7,6	0,0	21,3	27 152 349
17	Germany	20,2	1,0	0,0	0,1	21,2	17 472 000
18	Austria	11,4	6,6	0,0	0,6	18,6	1 543 518
19	Spain	13,3	3,6	0,0	0,1	17,0	7 483 790
20	Other OECD						36 521 821
	OECD	**11,6**	**5,4**	**1,4**	**0,3**	**18,8**	**221 020 786**

China (> 50M BB connections), other Asian tigers, etc.

OECD BB subs per 100 inhabitants



Conclusions on BB penetration

- Subscriptions are growing at 20+% per annum
  - developed countries like to talk about connections per inhabitants. BB is however mostly a service per household. E.g. in Korea households have more people than in the West.
- Aging of ISDN/PSTN technology may be replaced by Multiservice Access Nodes with PSTN/ISDN/xDSL/… line cards + Next Generation Network backbone providing all services from the same platform and cutting costs.
  - IMS is the signaling and services machinery used in NGN.
- Nrof of Fiber connections to homes are gaining market share. BB radio is also gaining a limited success while xDSL and Cable dominate the BB access market.
  - in Korea xDSL penetration is declining – replacement by Fiber is taking place: nrof of Fiber homes has more than doubled between June 2006 and June 2007.
  - In Japan xDSL is also absolutely declining, replacement by Fiber is taking place.

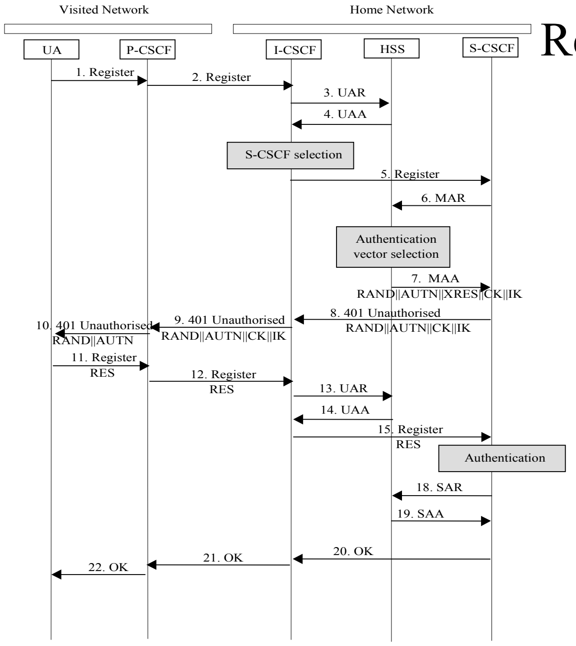
Conclusions on SIP

- SIP is a native IP-network signaling system suitable for Broadband networks
  - Needs compression when used e.g. in shared media cellular networks such as 3G WCDMA
  - Also, cellular networks of the future are going to be BB networks
- Most signaling and service architecture development in the world now is SIP oriented
  - Several IETF groups are producing a broad set of documents related to SIP, 3GPP has produces >> n x 100 pages…
  - SIP architecture = base protocol + extensions
  - Recent developments include conferencing, emergency services, PSTN/ISDN emulation (TISPAN), Peer-to-Peer SIP etc.
- Deployment
  - BT NGN is based on SIP and IMS and will replace BT's PSTN in a few years
  - No attractive services in cellular networks so far based on IMS. Due to well working CS services, operators are not in a hurry to replace CS services with packet based IMS produced services in cellular networks. New attractive services are needed.

Appendix B – 3GPP IMS call flows

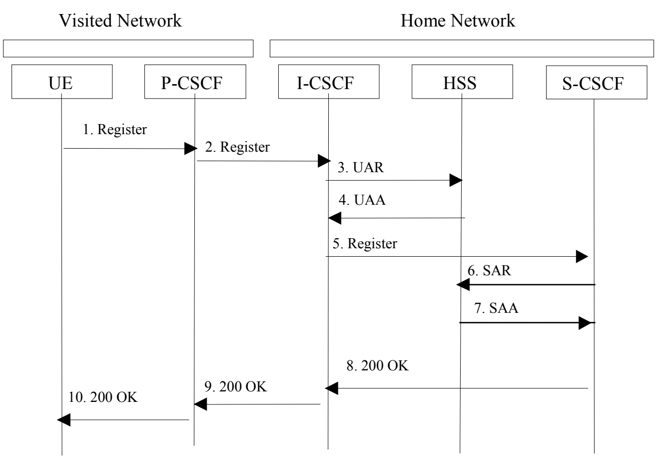
Registration – user not registered

Source: 29.228 v 7.0.0



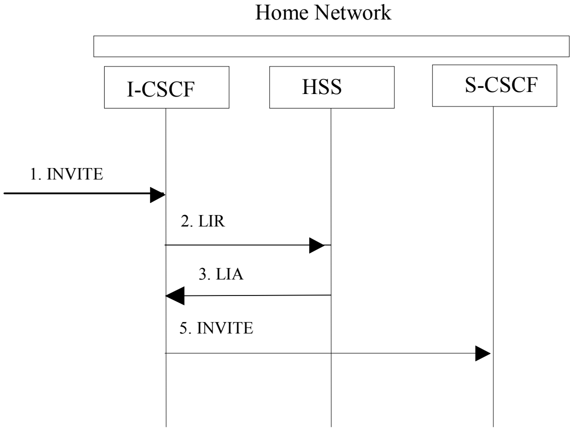
Registration – user currently registered

Source: 29.228 v 7.0.0

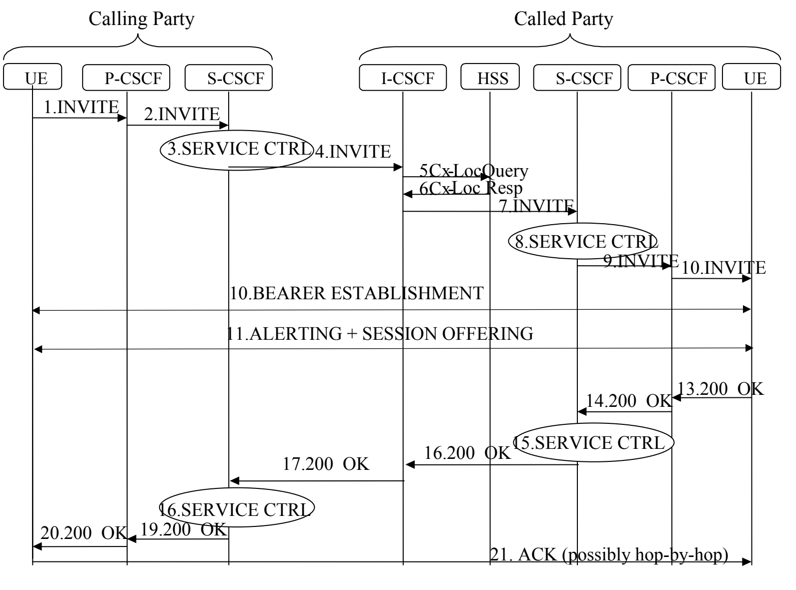


Mobile Terminated Session Setup

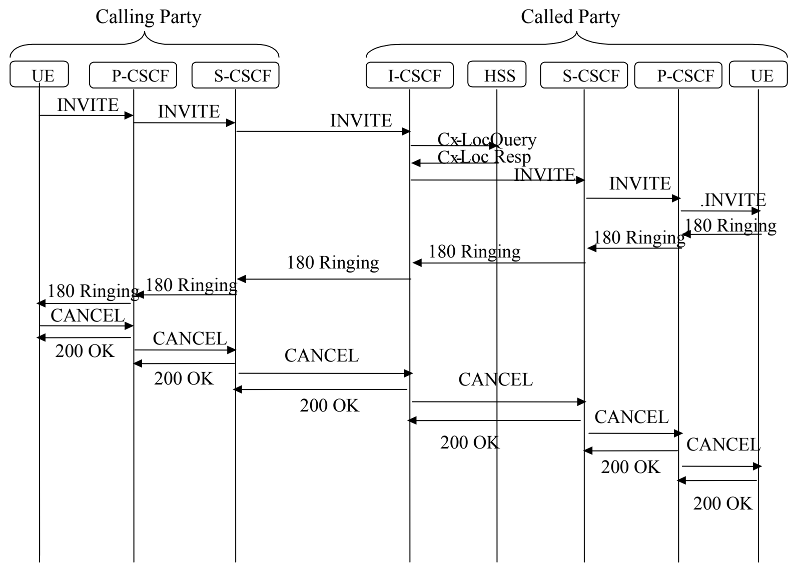
Source: 29.228 v 7.0.0



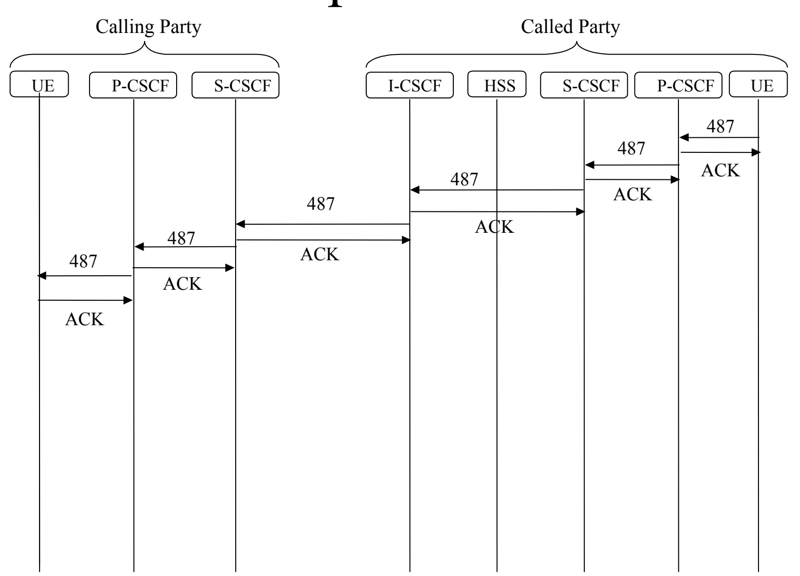
Mobile to Mobile Call



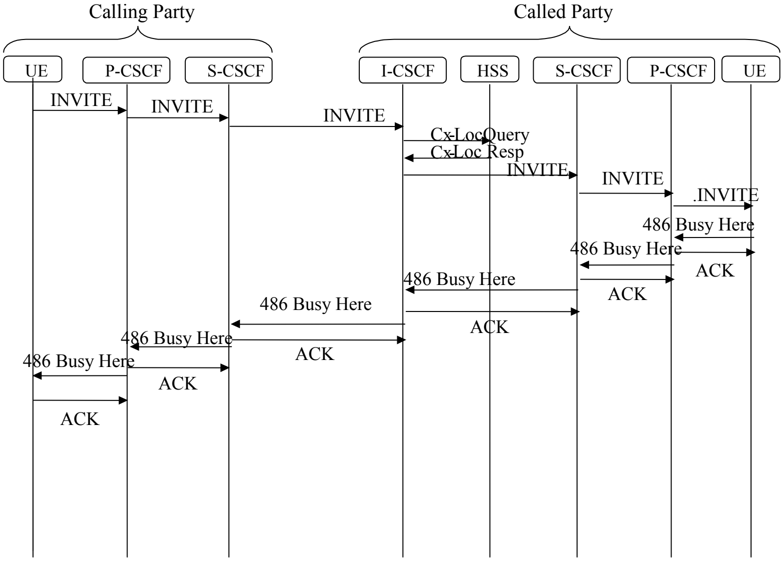
Call flow examples 1. - no answer



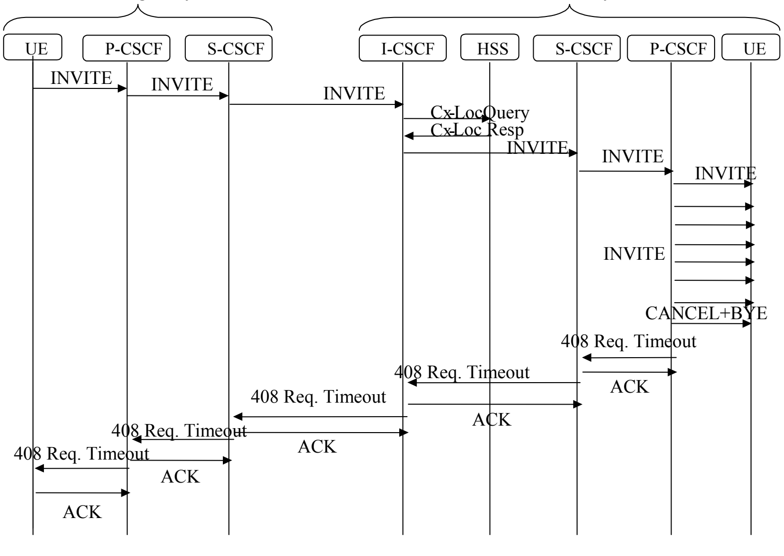
Call flow examples 1. - no answer 2.



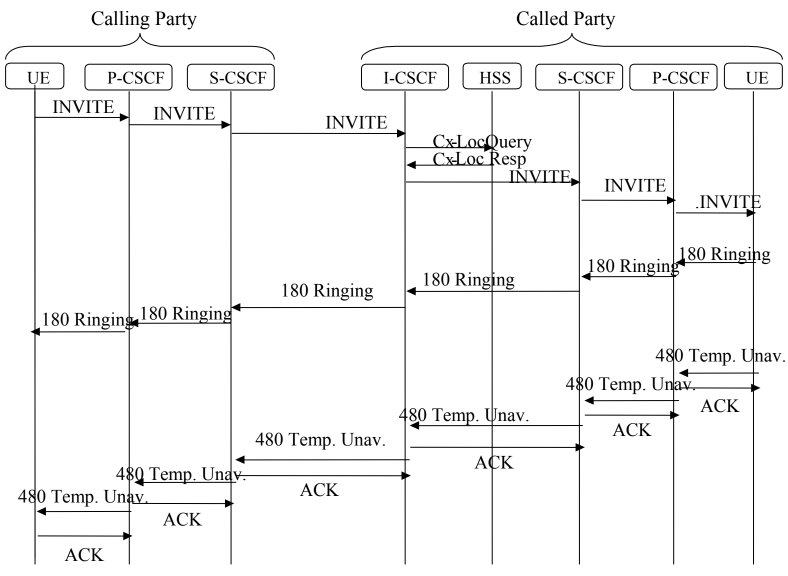
Call flow examples 2. - busy

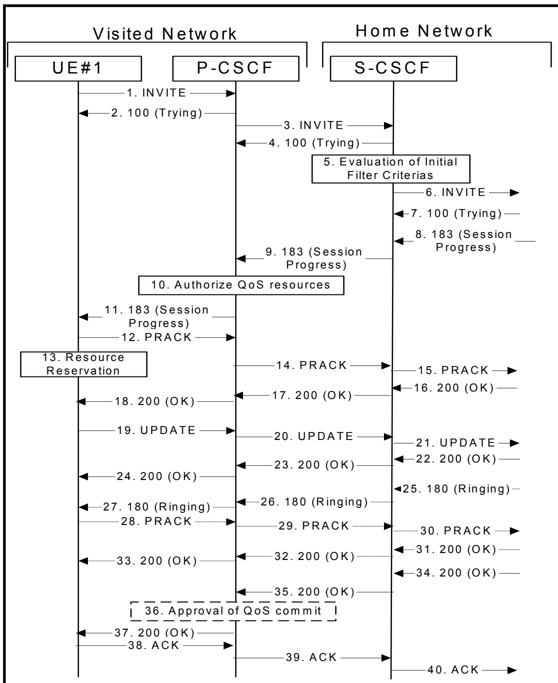


Call flow examples 3. - no response



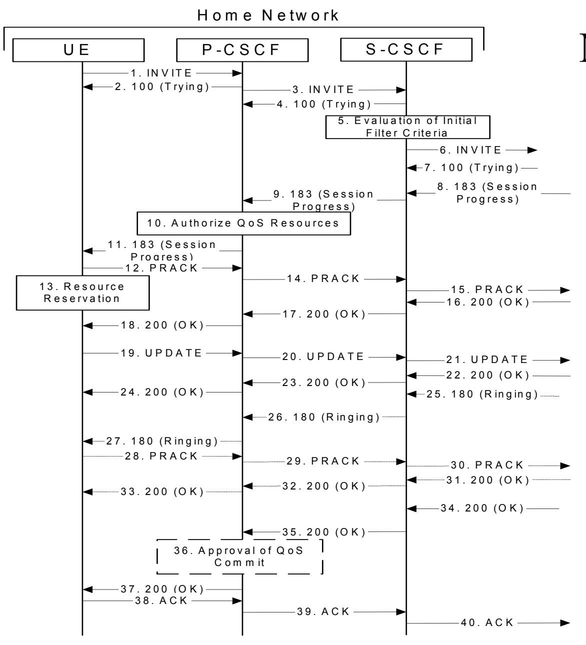
Call flow examples 4. - temporarily unavailable





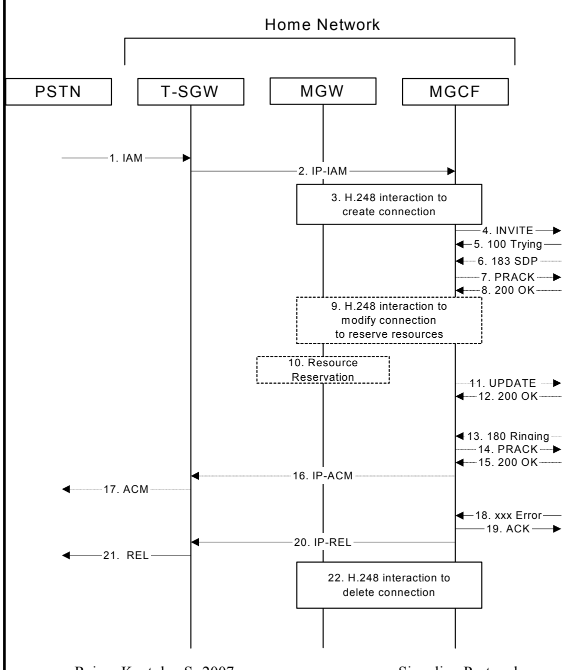
MO session setup - roaming

24.228 v 5.1.4



MO Session setup – user in home network

24.228 v 5.1.4



PSTN originated session

24.228 v 5.1.4