Session Initiation Protocol (SIP) and 3GPP IMS and Their roles in Enabling NEW services

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Acknowledgements

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- Christophe Gourraud, The IMS Lantern, http://theimslantern.blogspot.com
- 3GPP, http://www.3gpp.org
- IETF, http://www.ietf.org
- OMA, http://www.openmobilealliance.org
- Tech-Invite, http://tech-invite.com
- Gonzalo Camarillo, Miguel A. Garcia-Martin, "The 3G IP Multimedia Subsystem (IMS)," 2nd Edition, Wiley, 2006.
- Mikka Poikselka et al, "The IMS IP Multimedia Concepts and Services," 2nd Edition, Wiley, 2006.
- Musa Unmehopa et al, "Parlay/OSA: from Standards to Reality,", Wiley, 2006.
- T. Magedanz, http://www.fokus.fraunhofer.de/ngni
- Kimmo Raatikainen, http://www.cs.helsinki.fi/Kimmo.Raatikainen
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- J.C. Chen, T. Zhang, "IP-based Next-Generation Wireless Networks," Wiley.
- Jochen Schiller, "Mobile Communications," 2nd Edition, Addison Wesley, http://www.inf.fu-berlin.de/inst/ag-tech/resources/mobile_communications.htm
- Cornelia Kappler, http://www.tkn.tu-berlin.de/curricula/ws0304/vl-umts/
- Leo Petrak, Christian Hoene, Georg Carle, http://net.informatik.uni-tuebingen.de/de/lehre/umts-voip/ss2006/
- Wei-Kuo Chiang, http://www.cs.ccu.edu.tw/~wkchiang
- Vishal Kumar Singh, Henning Schulzrinne, http://www1.cs.columbia.edu/~vs2140/presence
- Hannes Tschofenig, http://www.tschofenig.com/wp/wp-content/uploads/2007/04/w3c_geopriv-slides.ppt
- DeWayne Sennett, UBC-IEEE Workshop on Future Wireless Systems http://bul.ece.ubc.ca/IMS_BUL06.pdf
- C.V. Chakravathy, "IP Multimedia Subsystems: A tutorial", IMSexpo, Oct 2006.
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Outline

- SIP Overview and the vision of a SIP-based Service Architecture

- An example of SIP's service enabling role in the 3GPP Internet Multimedia Subsystem (IMS)
  - What is IMS ?
  - Realization via 3GPP IMS
    - Functional Overview of IMS
    - Service/Application Layer of IMS
      - IMS Service Profile Routing

SIP

- Session Initiation Protocol
- Comes from IETF

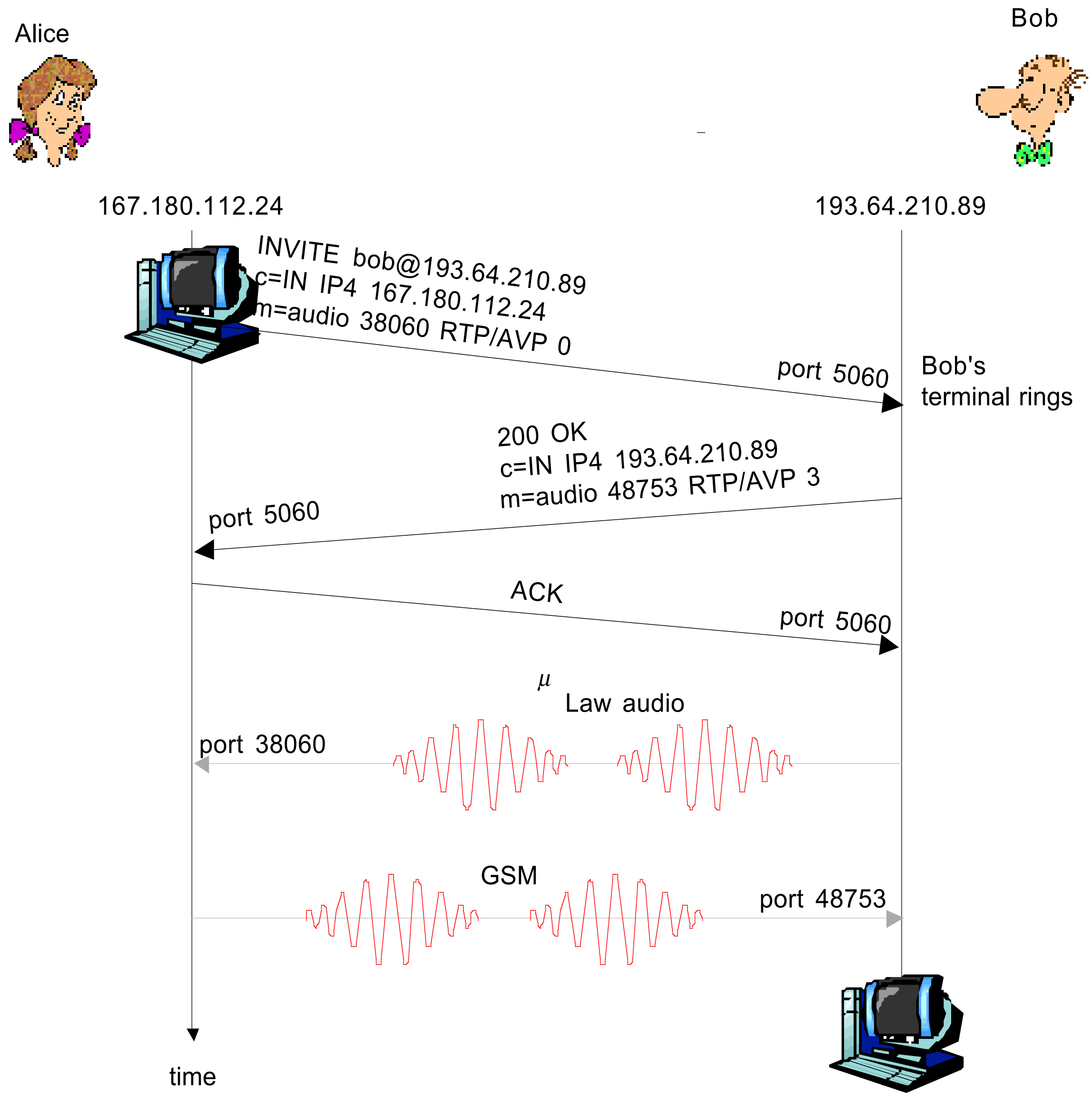
<u>SIP long-term vision</u>

- All telephone calls and video conference calls take place over IP
- People are identified by names or e-mail addresses, rather than by phone numbers.
- You can reach the callee, no matter where the callee roams, no matter what IP device the callee is currently using.

SIP Services

- Setting up a call
  - Provide mechanisms for caller to let callee know she wants to establish a call
  - Provide mechanisms so that caller and callee can agree on media type and encoding.
  - Provide mechanisms to end call.
- Determine current IP address of callee.
  - Maps mnemonic identifier to current IP address
- Call management
  - Add new media streams during call
  - Change encoding during call
  - Invite others
  - Transfer and hold calls

Setting up a call to a known IP address



- Alice's SIP invite message indicates her port number & IP address. Indicates encoding that Alice prefers to receive (PCM ulaw)
- Bob's 200 OK message indicates his port number, IP address & preferred encoding (GSM)
- SIP messages can be sent over TCP or UDP; here sent over RTP/UDP.
- Default SIP port number is 5060.

Setting up a call (more)

- Codec negotiation:
  - Suppose Bob doesn't have PCM ulaw encoder.
  - Bob will instead reply with 606 Not Acceptable Reply and list encoders he can use.
  - Alice can then send a new INVITE message, advertising an appropriate encoder.
- Rejecting the call
  - Bob can reject with replies "busy," "gone," "payment required," "forbidden".
- Media can be sent over RTP or some other protocol.

Example of SIP message

```
INVITE sip:bob@domain.com SIP/2.0
Via: SIP/2.0/UDP 167.180.112.24
From: sip:alice@hereway.com
To: sip:bob@domain.com
Call-ID: a2e3a@pigeon.hereway.com
Content-Type: application/sdp
Content-Length: 885

c=IN IP4 167.180.112.24
m=audio 38060 RTP/AVP 0
```

Notes:

- HTTP message syntax
- SDP = session description protocol
- Call-ID is unique for every call.

- Here we don't know Bob's IP address. Intermediate SIP servers will be necessary.
- Alice sends and receives SIP messages using the SIP default port number 5060.
- Alice specifies in Via: header that SIP client sends and receives SIP messages over UDP

Name translation and user location

- Caller wants to call callee, but only has callee's name or e-mail address.
- Need to get IP address of callee's current host:
  - user moves around
  - DHCP protocol
  - user has different IP devices (PC, PDA, car device)
- Result can be based on:
  - time of day (work, home)
  - caller (don't want boss to call you at home)
  - status of callee (calls sent to voicemail when callee is already talking to someone)

<u>Service provided by SIP servers:</u>

- SIP registrar server
- SIP proxy server

SIP Registrar

- When Bob starts SIP client, client sends SIP REGISTER message to Bob's registrar server

(similar function needed by Instant Messaging)

Register Message:

```
REGISTER sip:domain.com SIP/2.0
Via: SIP/2.0/UDP 193.64.210.89
From: sip:bob@domain.com
To: sip:bob@domain.com
Expires: 3600
```

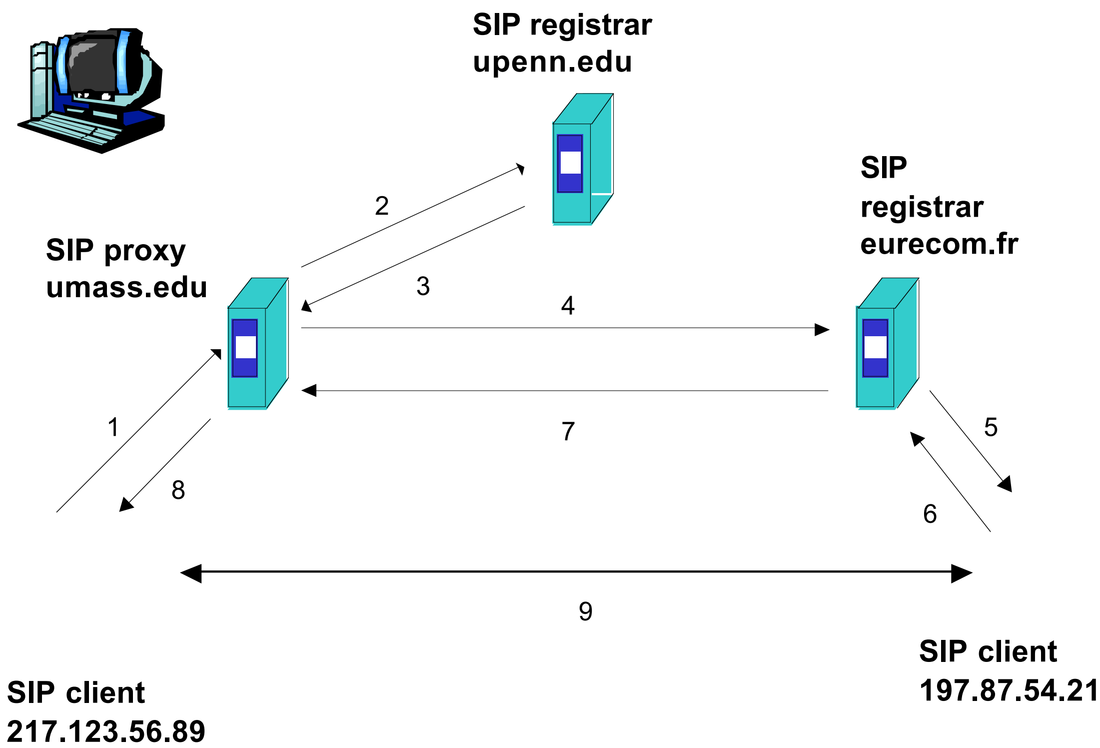
SIP Proxy

- Alice sends invite message to her proxy server
  - contains address sip:bob@domain.com
- Proxy responsible for routing SIP messages to callee
  - possibly through multiple proxies.
- Callee sends response back through the same set of proxies.
- Proxy returns SIP response message to Alice
  - contains Bob’s IP address

- Note: proxy is analogous to local DNS server

Example

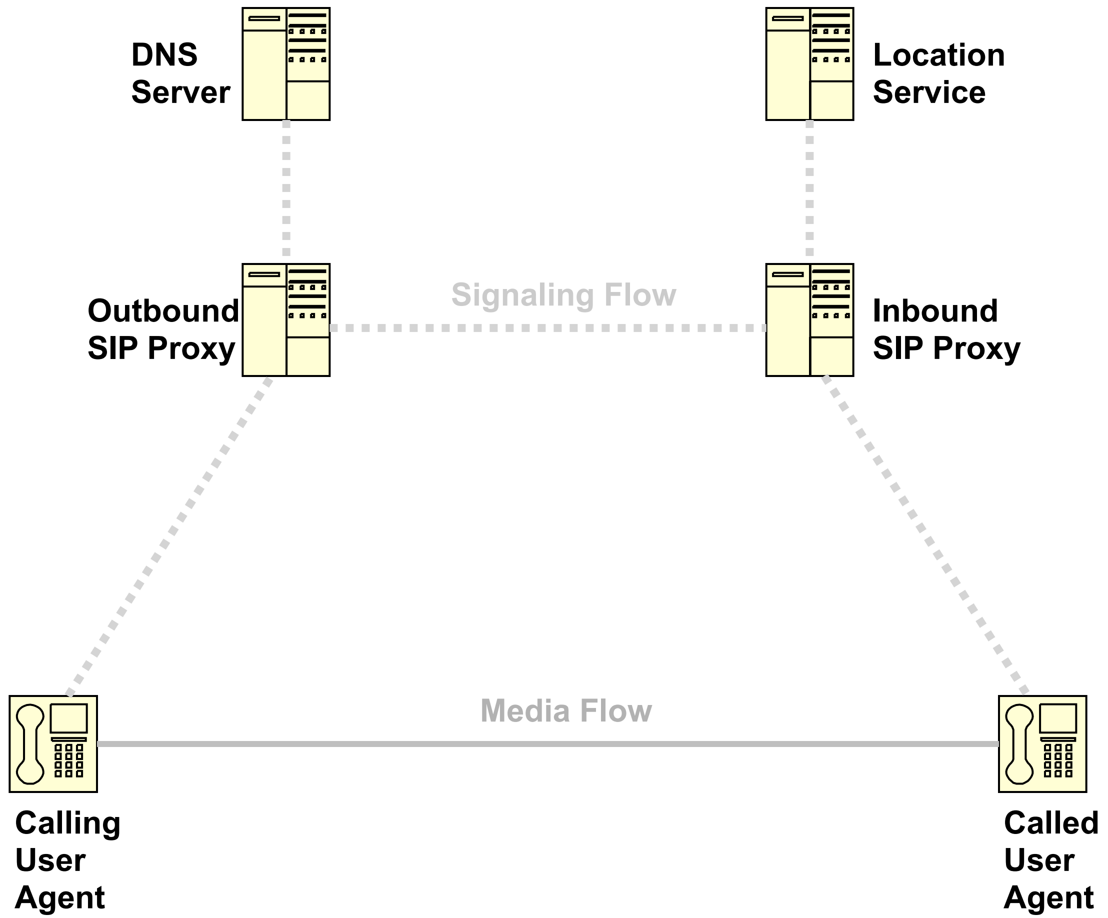
Caller jim@umass.edu with places a call to keith@upenn.edu



(1) Jim sends INVITE message to umass SIP proxy.
(2) Proxy forwards request to upenn registrar server.
(3) upenn server returns redirect response, indicating that it should try keith@eurecom.fr

(4) umass proxy sends INVITE to eurecom registrar.
(5) eurecom registrar forwards INVITE to 197.87.54.21, which is running keith's SIP client.
(6-8) SIP response sent back
(9) media sent directly between clients.
Note: also a SIP ack message, which is not shown.

SIP Trapezoid

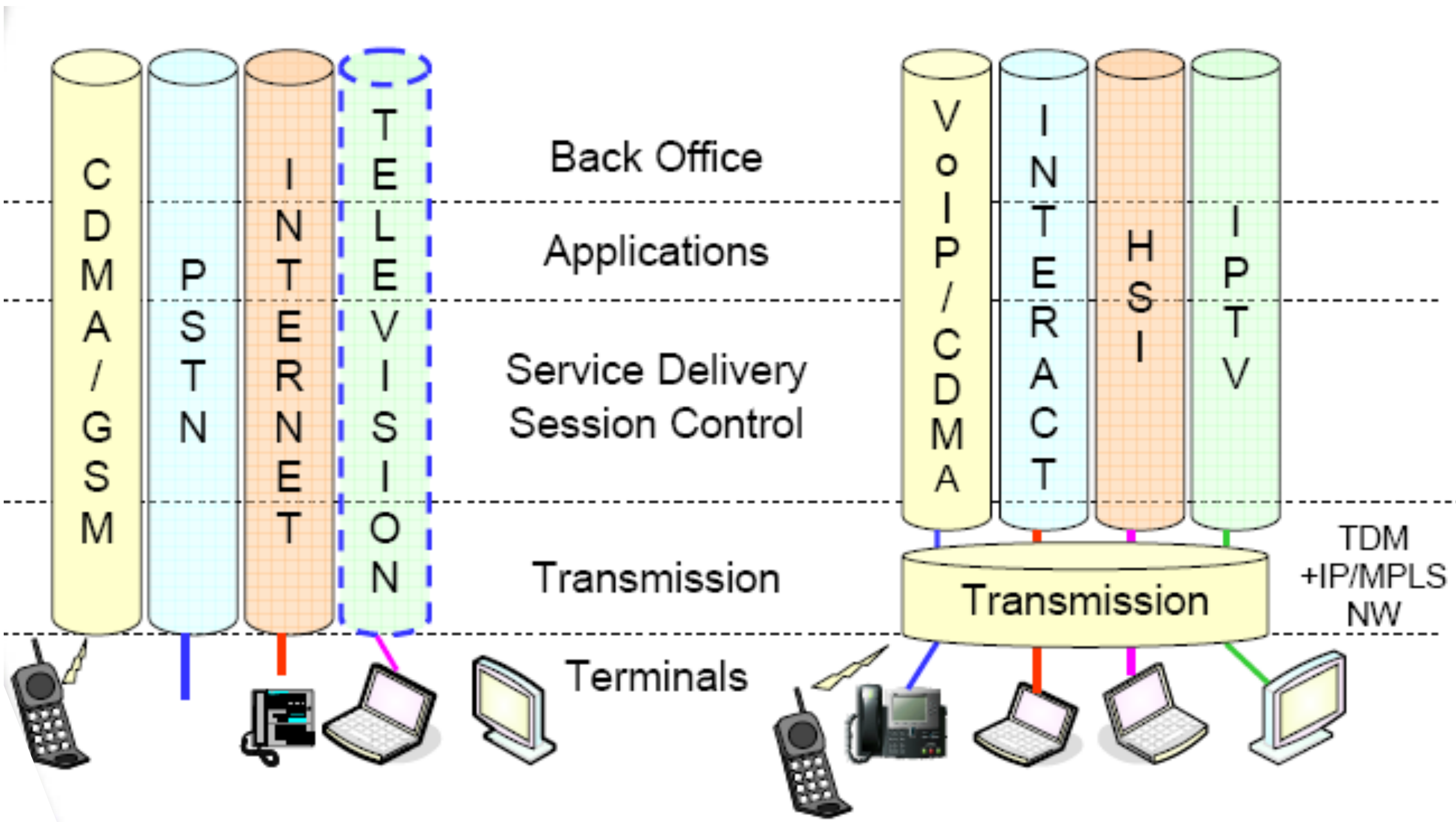


More SIP References

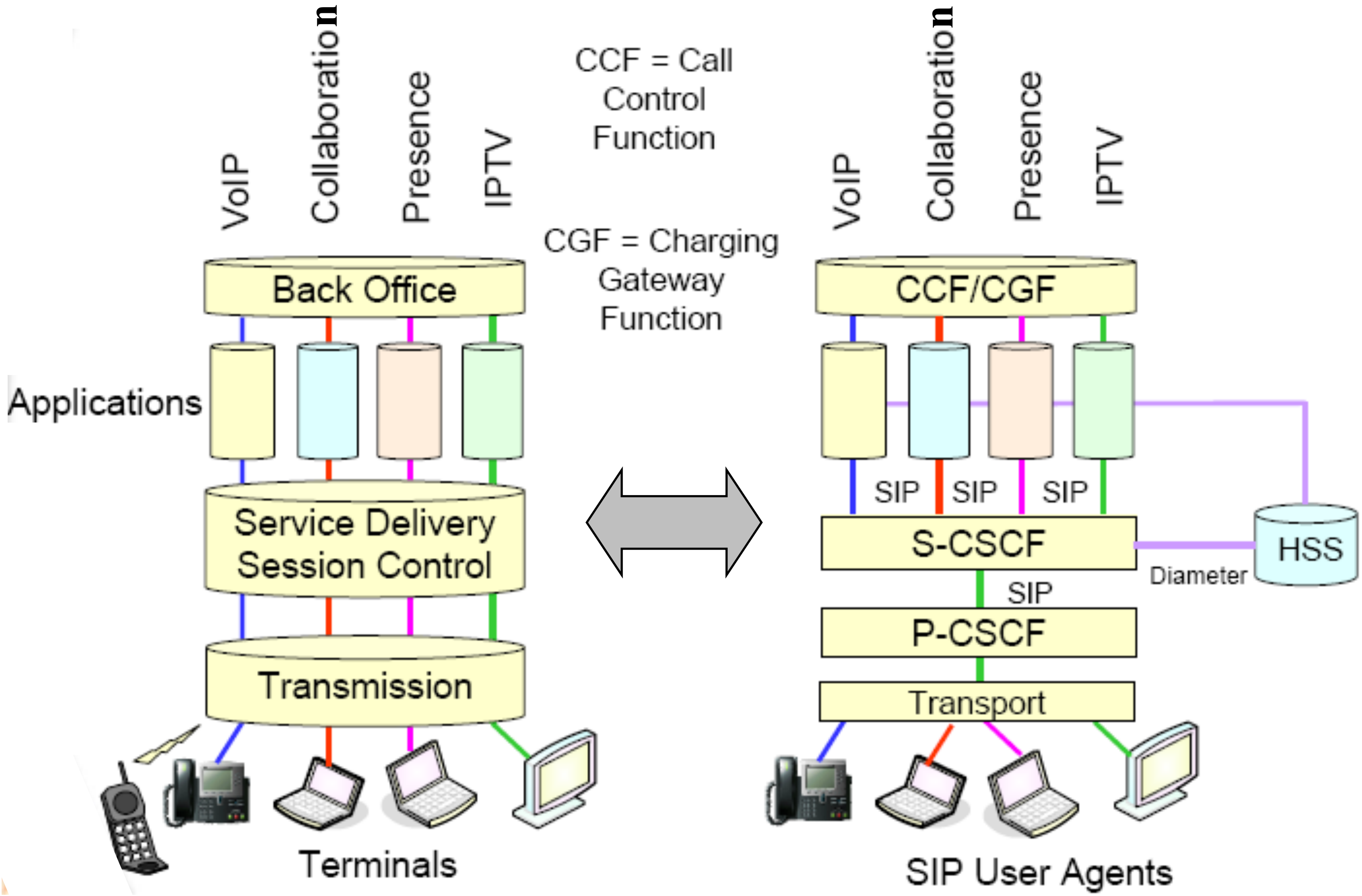
- Jonathan Rosenberg, “A Hitchiker’s Guide to the Session Initiation Protocol (SIP)”, IETF RFC 5411, Jan. 2009.
- A. Johnston, Ed., “Session Initiation Protocol Services Examples”, IETF RFC 5359, Oct. 2008.
  - for full coverage with examples on how to use of SIP to support IP Centrex /PBX applications
- Jonathan Rosenberg, “A Framework for Conferencing with the Session Initiation Protocol (SIP),” IETF RFC4353 (informational), Feb. 2006.

The 3GPP IP Multimedia Subsystems (IMS) and its realization via the Session Initiation Protocol (SIP) Framework

"Limited" Convergence Models in Today's Networks



IMS: A Horizontal Service Enabling Platform

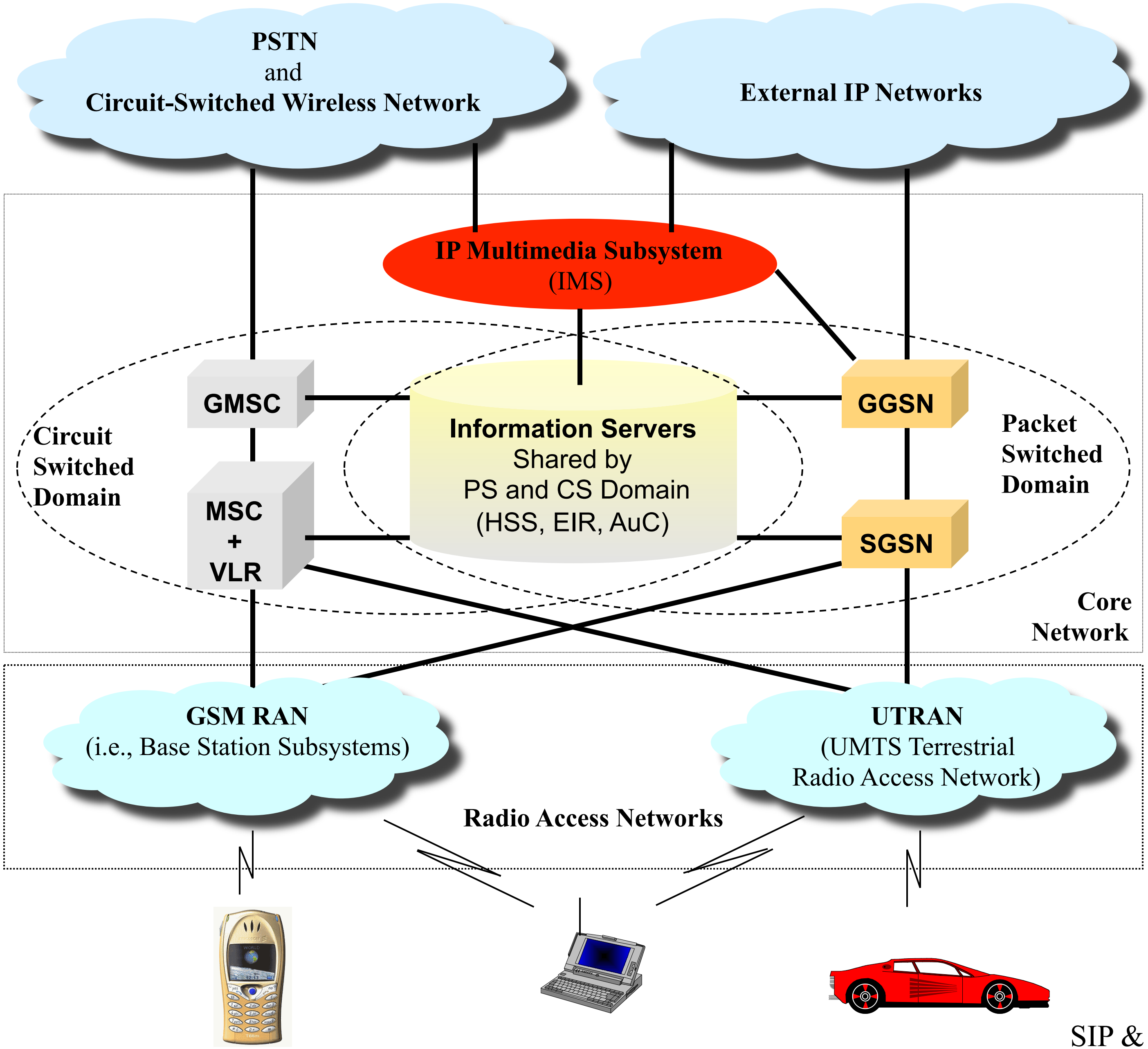


What is the IMS ?

- The IMS enables the provisioning of
  - IP-based Multimedia services
  - Other value added services, e.g.
    - Presence, Location-awareness, Group Management, Instant Messaging

    by providing a SINGLE Horizontal (shared) platform for new service creation
- While such services can also be offered without IMS, however,
  - The IMS provides a standardized, convenient and commercially viable way for offering them
    - In particular, IMS establish a "Call-control framework" to allow the Cellular/ Network Service Providers to keep service-intelligence inside their networks, and thus can provide QoS guarantees and charge for it, (as opposed to let end-points handle everything while becoming a dumb-fat-pipe itself).
    - Seen as the "Walled Garden" for the Mobile Service Provider by some ?
- The IMS can, in principle, be operated by a 3rd party, i.e. not the Network Service Provider but unlikely in practice
- IMS thus offers Network Intelligence (Service) access to Millions of 3rd Party application developers (IT/Web-based-oriented)

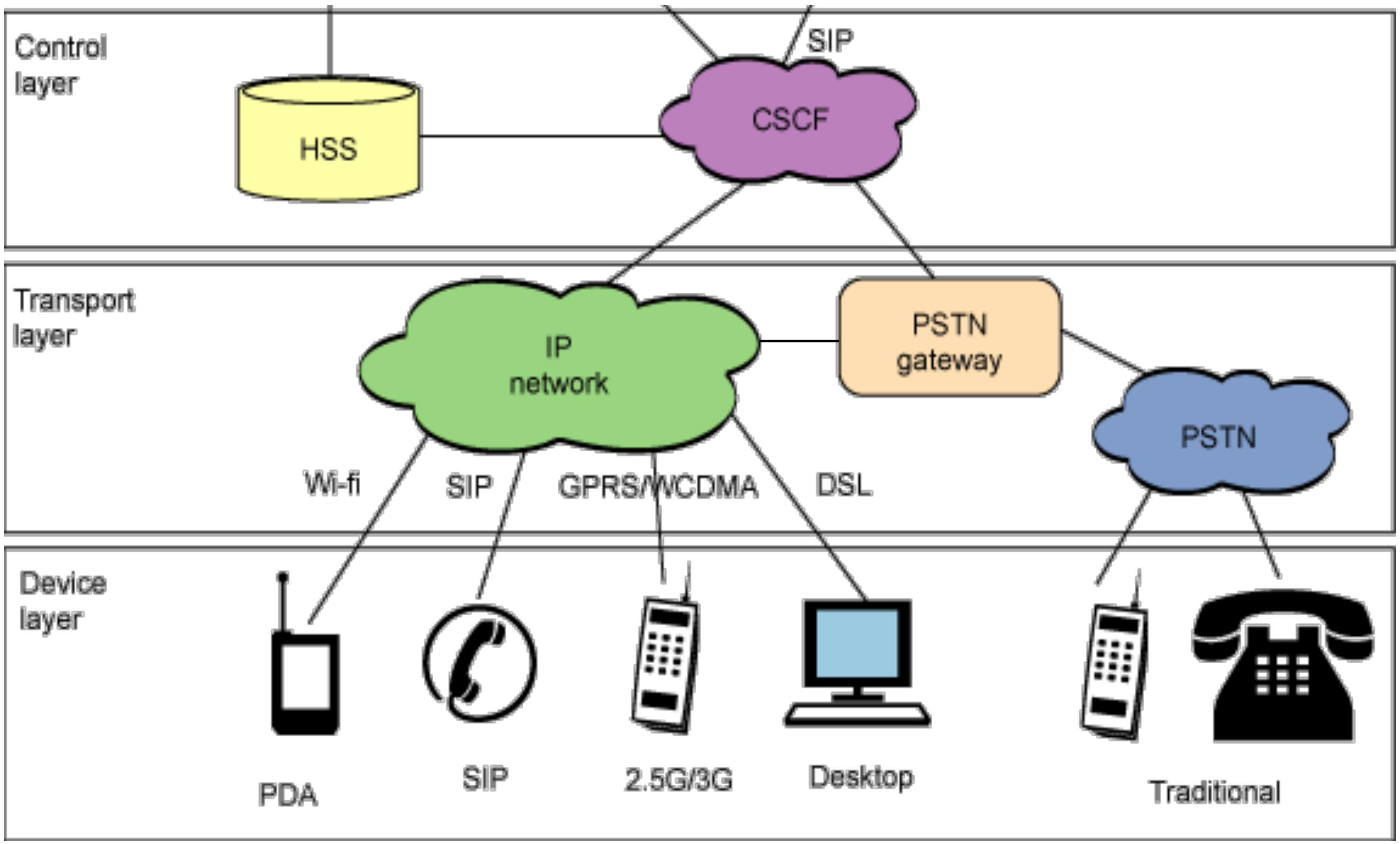
3GPP conceptual network architecture (Release 5)



What is the IMS ? (cont'd)

- In 3G Cellular networks, the IMS uses the Radio Access Networks (RAN) and Packet-Switched-domain for
  - transporting of user and signaling traffic and
  - taking care of Terminal Mobility
  - The IMS can be deployed without a Circuit-Switched-(CS) domain but IMS can enrich CS services as well, e.g. Voice Call Continuity.
- The IMS Call-control framework has also become a key vehicle for the integration/ interworking of Next Generation Wired and Wireless Public Networks
  - Fixed Mobile Convergence (FMC)
  - Interworking of different Radio Access Technologies networks
- Interfaces and protocols conform as much as possible with IETF Specifications:
  - Re-use IETF protocols (SIP/SDP/RTP/Diameter/etc.)

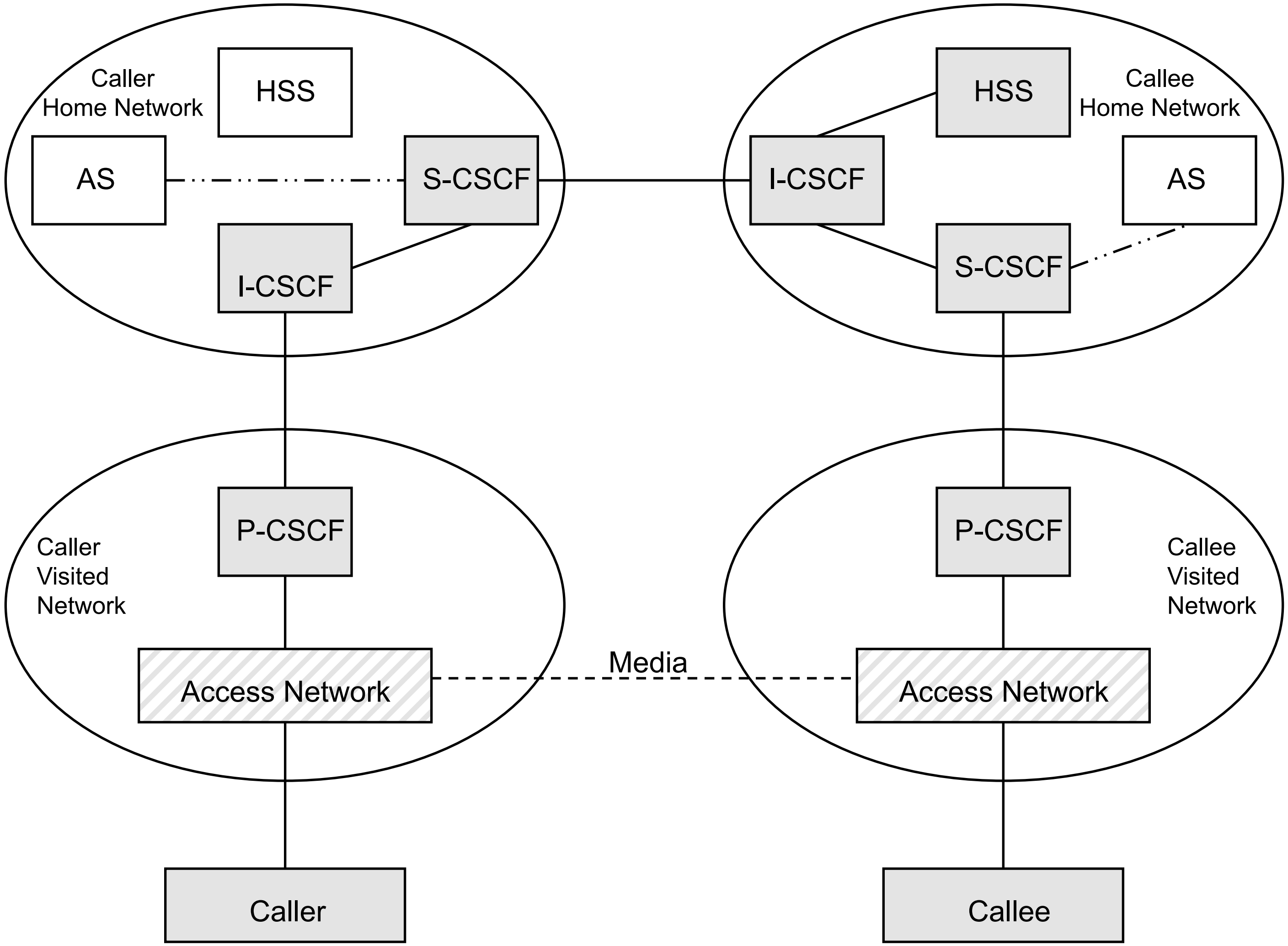
Interworking between VoIP, PSTN and Cellular Services via the (SIP-based) IMS



The Realization of IMS via IETF Protocols

- Both IMS signaling and media are transported over IP.
- IETF suite of Internet Multimedia Protocols are used:
  - Session Initiation Protocol (SIP)
  - Session Description protocol (SDP)
  - Real Time Transfer Protocol (RTP)
- SIP is the signaling protocol for IMS session establishment, and for Presence and Instant Messaging.

The Realization of IMS using SIP



Under IMS terminology,
SIP proxy/registrar servers becomes *-CSCF (Call Session Control Function)

IMS Functional Entities

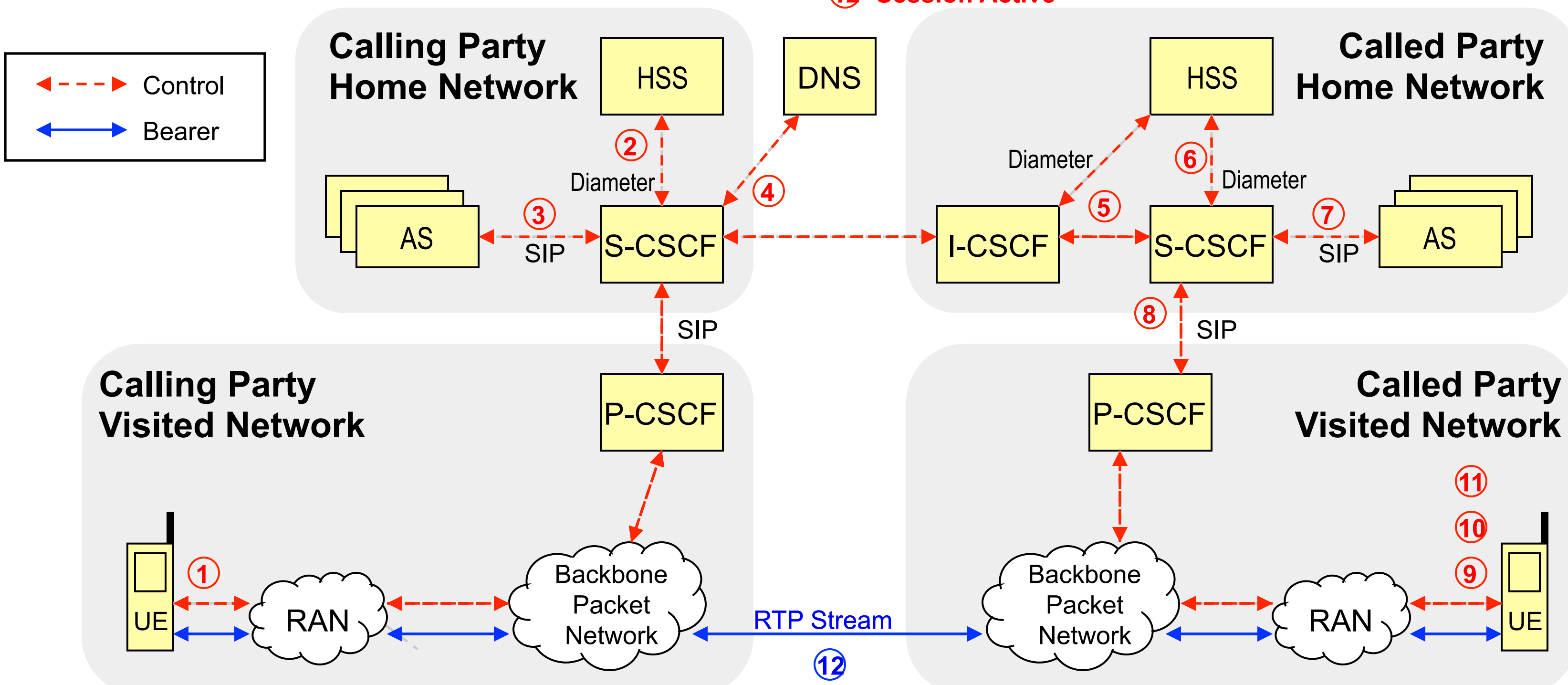
- Home Subscriber Server (HSS): extension of HLR to include the data pertaining to the IP Multimedia Subsystem.
  - Access from the CSCF will be based on IETF protocols (Diameter)
- Call/Session Control Function (CSCF) - provides call control functions :
  - Serving CSCF
    - SIP registrar, with cooperation from HSS
    - Session control call state machine
    - Interaction with service platforms for service control, with service triggers
      - Support the so-called IMS service-profile based routing
  - Proxy CSCF
    - SIP proxy server for the mobile, acting on behalf of UE within IMS
    - Maintain security association with the mobile
    - Policy control function for QoS authorization
  - Interrogating CSCF
    - Allocate the S-CSCF during IMS registration
    - Determine the S-CSCF for incoming calls
    - May hide network topology

Differences between IMS CSCFs and IETF SIP servers

- The IMS was designed based on RFC 3261 (the CSCF is basically also a SIP Server) but much more were added for IMS:
  - Subscriber Management, Service Control, Single-Sign-On User Authentication,
  - QoS/Media Authorization, Charging and Charging Correlation,
  - Resource Management, Interworking, Compression, Conferencing Support, Regulatory Service Support, etc.
- Most of the IMS functions were taken from the IETF or were afterwards defined in the IETF:
  - Update (RFC331), Preconditions (RFC3312), PRACK (RFC3262), Offer/Answer (RFC 3264),QoS/Media Authorization (RFC 3313), Event Notification (RFC 3265),Tel-URIs (RFC 2806), Service-Route (RFC3608), Asserted ID (RFC3325), DNS-Support (RFC 3263), SigComp (RFC3320, RFC3485, RFC 3486), ENUM (RFC2916, RFC2915), SIP Refer (RFC3515), Digest AKA (RFC 3310),Path-Header (RFC 3327), Security-Mechanism-Agreement (RFC3329), etc.
  - But there are some message type which are 3GPP-specific, those start with p* : 3GPP P-Headers (RFC3455) – an Informational RFC.
- A Service Infrastructure Network could also be built up starting with a standard RFC3261 SIP Server.
  - When extended to support the same Support Functions, then such a solution becomes similar to the IMS.

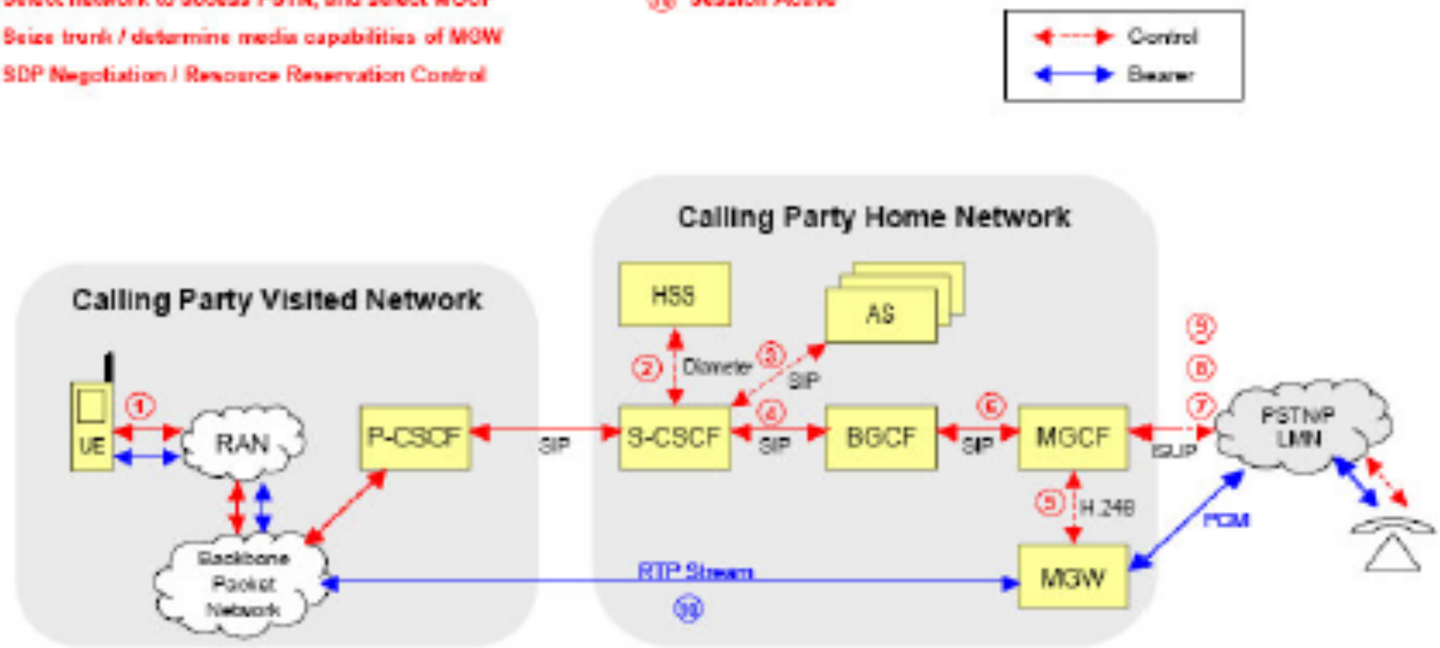
Sample call flow: IMS to IMS call

1. Initiate SIP Invitation
2. Retrieve Subscriber Profile (if needed)
3. Apply Service Logic
4. Retrieve Address of Called Party Home Network and Forward INVITE.
5. Identify Registrar of Called Party and Forward INVITE.
6. Retrieve Subscriber Profile (if needed)
7. Apply Service Logic
8. Forward INVITE to Called Party
9. SDP Negotiation / Resource Reservation Control
10. Ringing / Alerting
11. Answer / Connect
12. Session Active

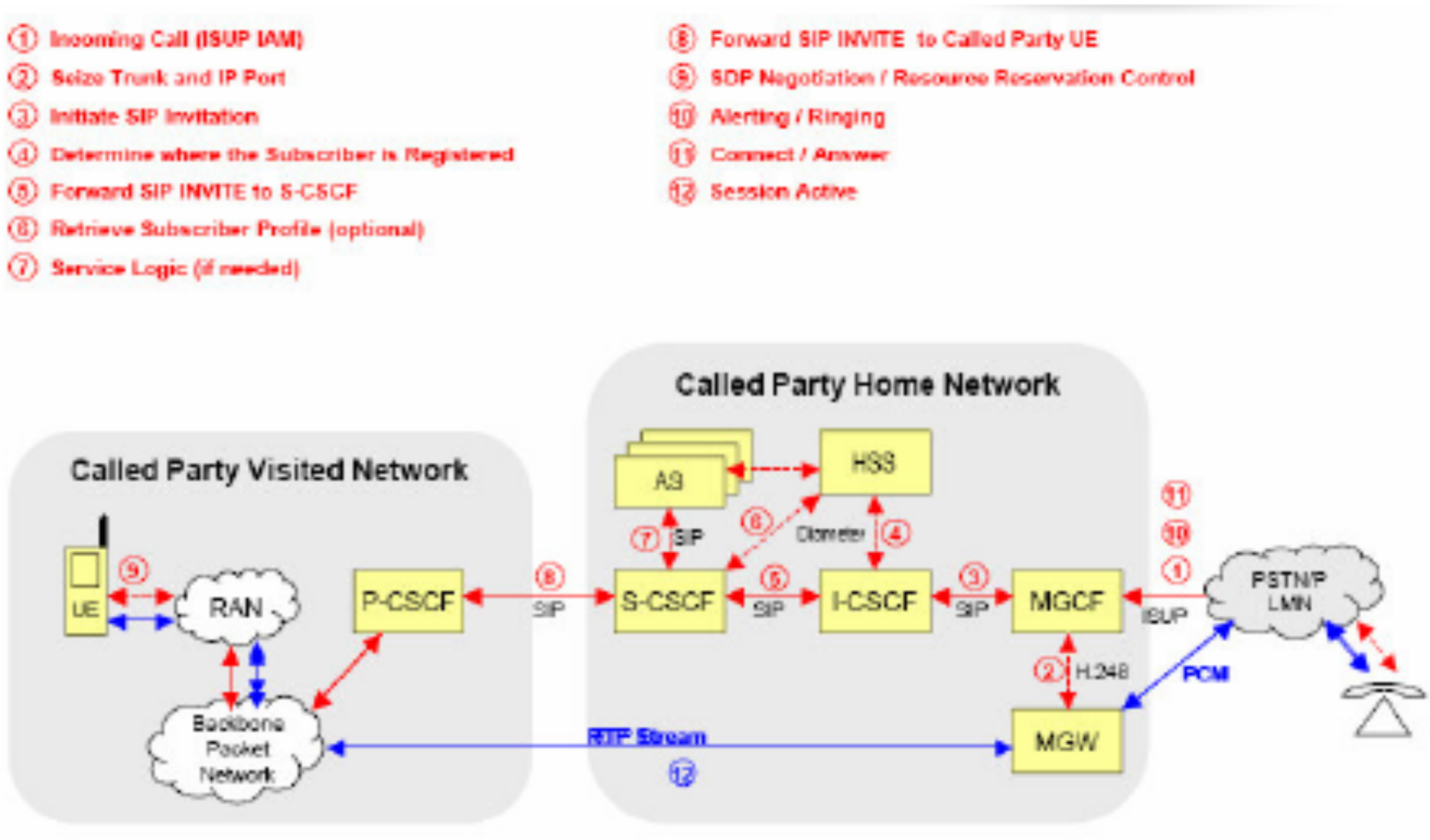


More Call Flow: IMS call to PSTN

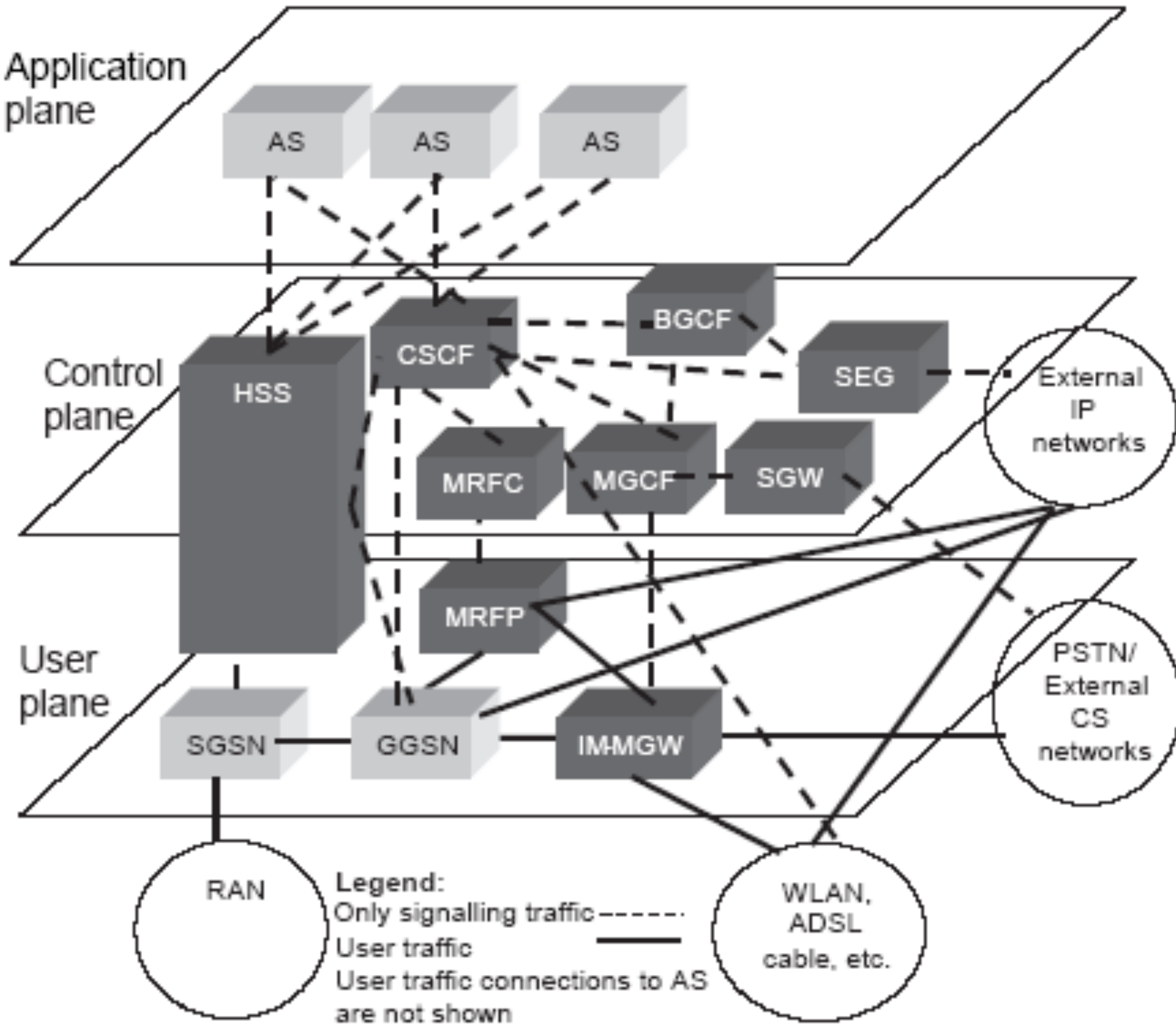
1. Initiate SIP Invitation
2. Retrieve Subscriber Profile (if needed)
3. Apply Service Logic
4. Select network to access PSTN, and select MGCF
5. Seize trunk / determine media capabilities of MGW
6. SDP Negotiation / Resource Reservation Control
7. ISUP IAM
8. Ringing / Alerting
9. Answer / Connect
10. Session Active



More Call Flow: PSTN call to IMS



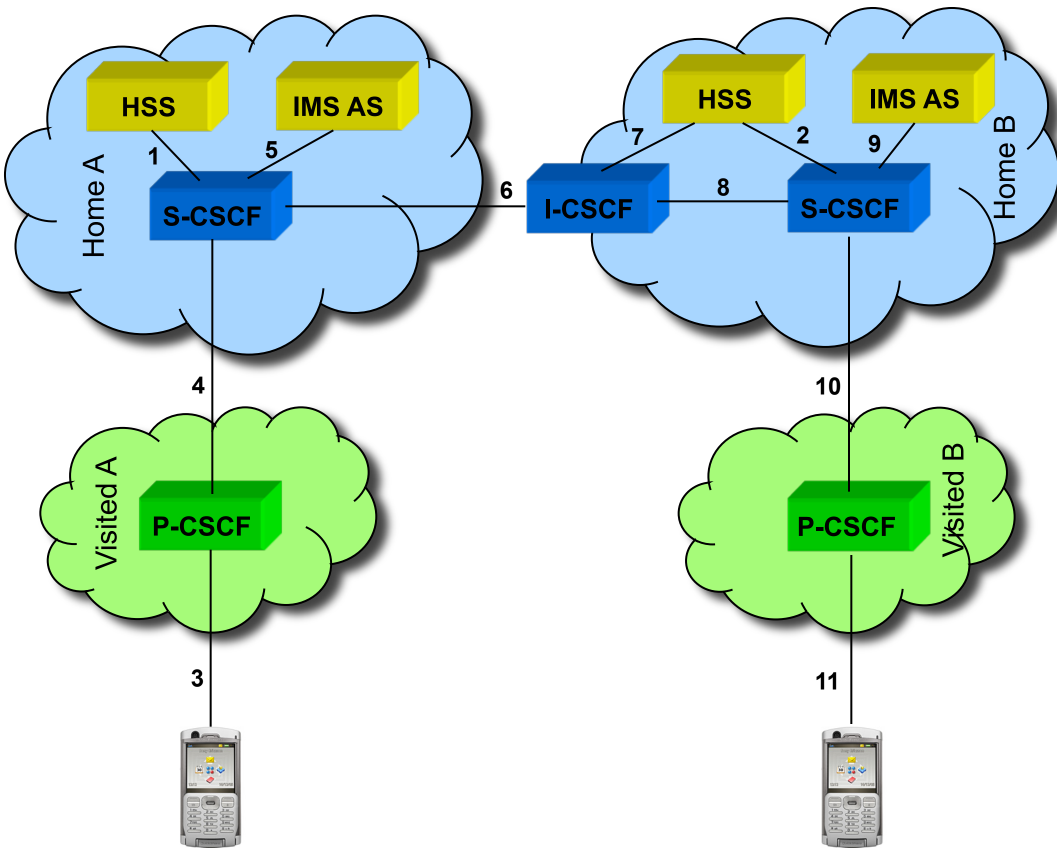
Simplified IMS and Layering Architecture



Other Components of IMS

- Media Gateway Control Function (MGCF) and IM Media Gateway (IM-MGW)
  - responsible for signaling and media interworking between PS and CS domains
- Multimedia Resource Function Processor (MRFP)
  - control the bearer on the M_b interface
  - process the media streams
- Multimedia Resource Function Controller (MRFC)
  - interpret signaling information from an S-CSCF or a SIP-based Application Server and control the media streams resources in the MRFP accordingly
  - Generate Charging Data Records (CDR)
- Breakout Gateway Control Function (BGCF)
  - select to which PSTN network a session should be forwarded
  - Forward the session signaling to the appropriate MGCF and BGCF in the destination PSTN network
- Security Gateway (SEG)
- Service Gateway (SGW)
- Application Server (AS): e.g. Application Server to provide location-based services or presence services

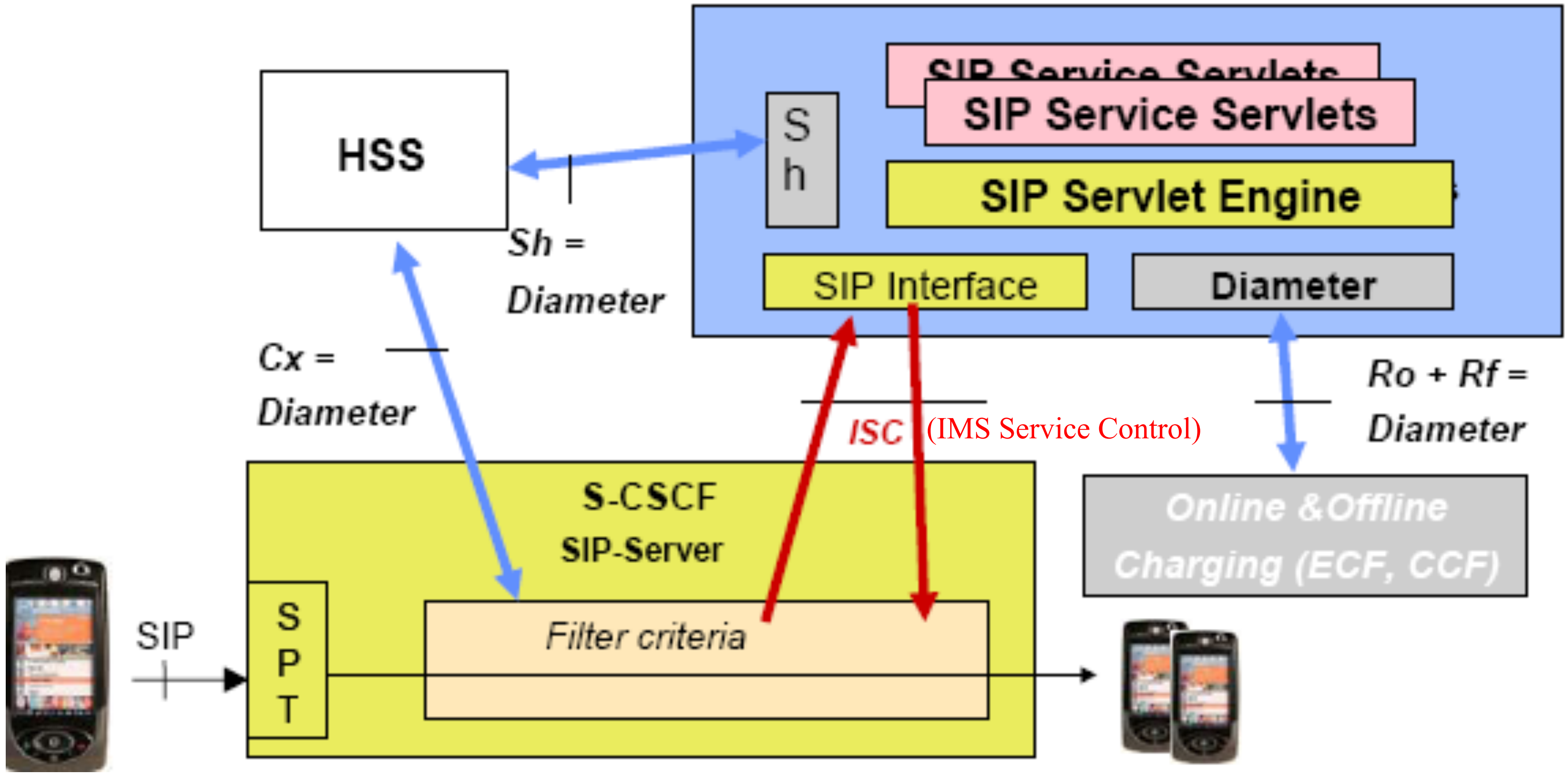
SIP-enabled IMS Service Routing



IMS Service Routing =
Service Profile based Routing

- In comparison to IETF SIP Routing where the originator of SIP request may specify a preferred path in the Route header, in IMS the P-CSCF removes this path and ensures that IMS SIP Routing is followed.
- SIP requests in IMS architecture are always routed to the Home S-CSCF, in both the originating and terminating network.
- The S-CSCF uses subscriber's Service Profile (downloaded during registration), to link-in the SIP AS which will process the SIP request.
- The Initial Filter Criteria (IFC) within the Subscriber Profile provide a simple service logic to decide which AS shall be linked-in. These rules are of static nature i.e. they do not change on a frequent basis.

SIP Application Server (Servlets)



An example of a Service Profile

John has a *service profile* associated to the sip:John@operator IMS Public User ID whose English meaning is:

(Priority 10): every registered originating INVITE for a voice session should be routed to sip:vcc_server@operator

(Priority 20): every registered originating INVITE should be routed to sip:multimedia_session_control_orig@operator

(Priority 25): every terminating INVITE should be routed to sip:multimedia_session_control_term@operator

(Priority 30): every terminating INVITE for voice should be routed to sip:vcc_server@operator

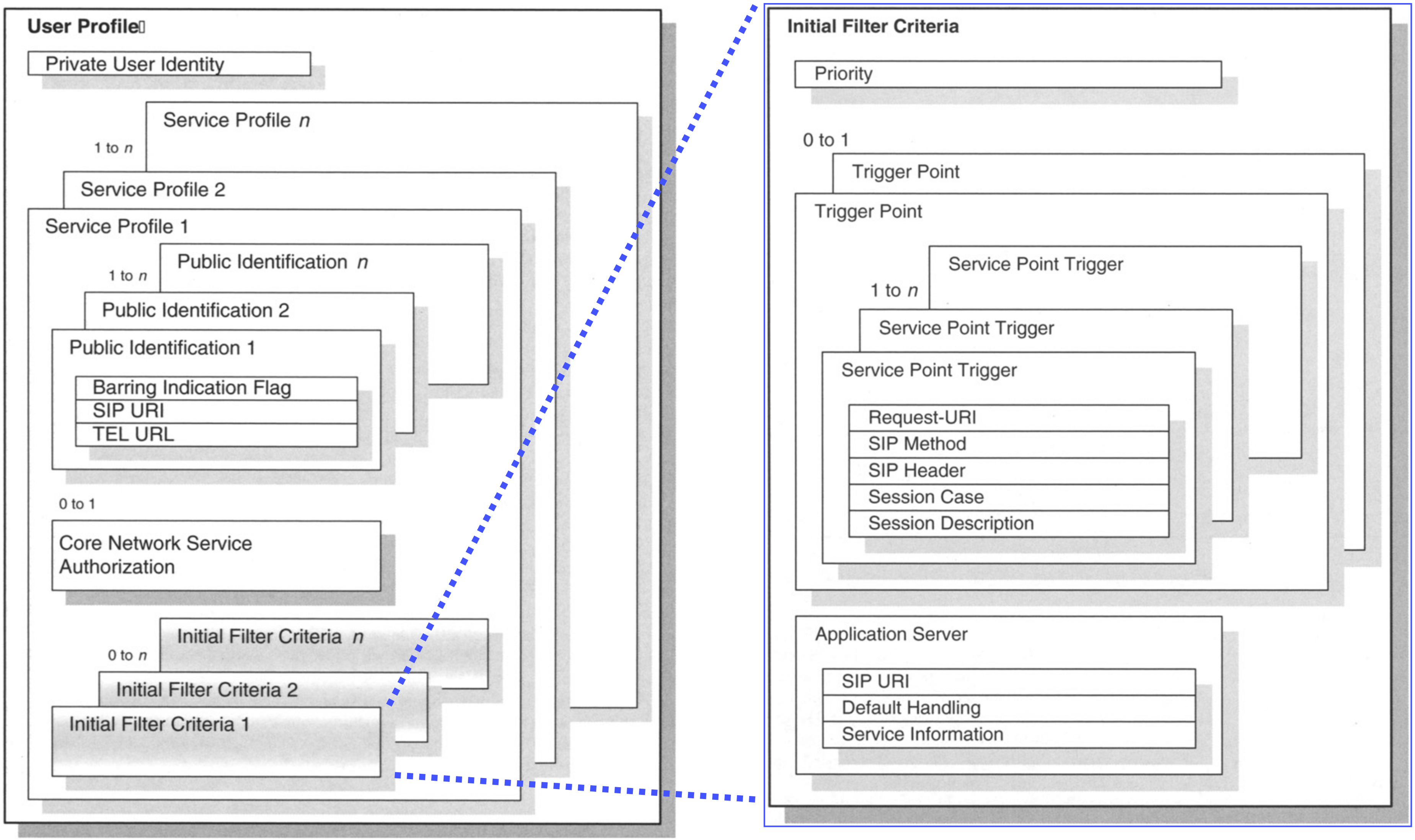
(Priority 32): every originating MESSAGE to request-URI sip:My_Family@operator should be routed to sip:message_exploder@operator

(Priority 40): every originating INVITE to request-URI sip:My_Family@operator should be routed to sip:multimedia_conference_server@operator

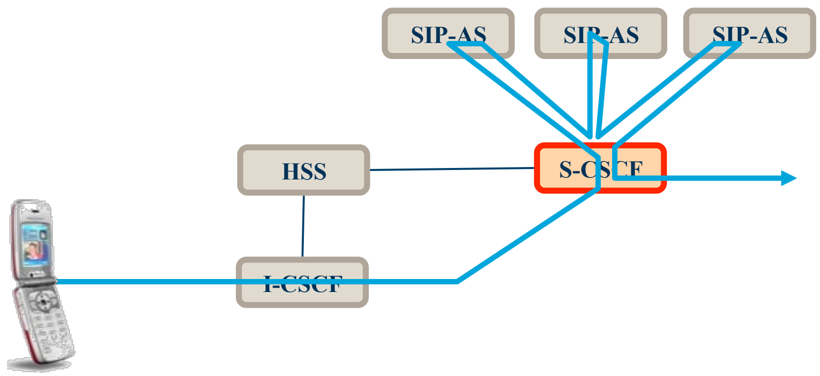
(Priority 50): every terminating SUBSCRIBE with a header called "event" whose value is "presence" should be routed to sip:presence_server@operator

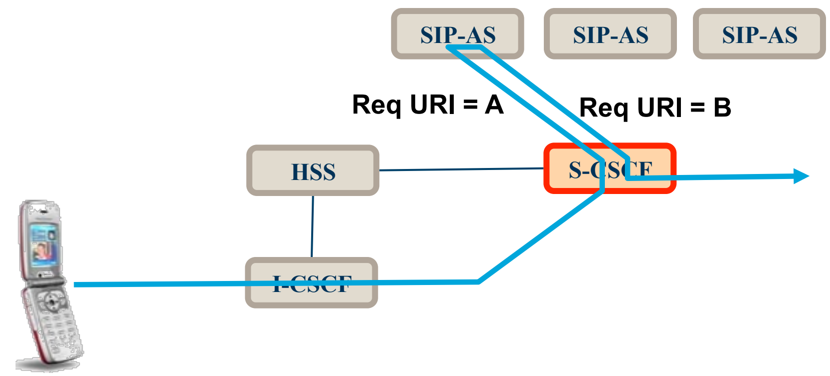
(Priority 60): every originating PUBLISH or SUBSCRIBE to Request-URI sip:John@operator with a header called "event" whose value is "presence" should be routed to sip:presence_server@operator

User Profile in IMS



The ISC Service Orchestration Models

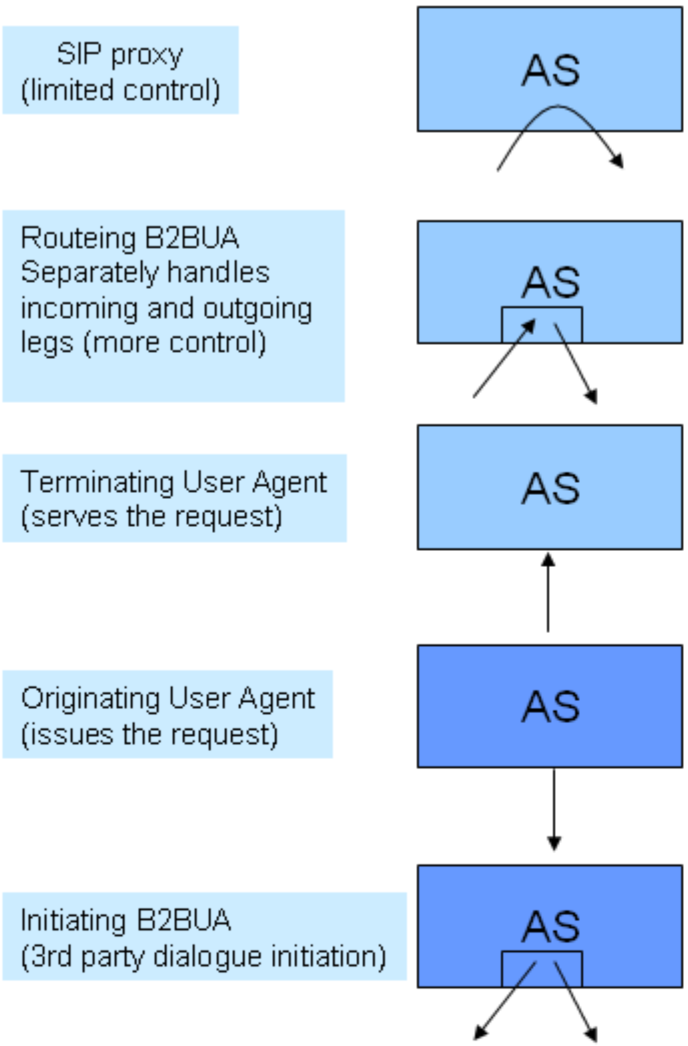




- The application server decides whether to remain linked-in for the whole session by adding its address to the Record-Route SIP header.
- Application Servers are unaware of the existence of other AS', and whether these will be linked-in.
- No service or session state will be passed between application servers unless they use proprietary extensions i.e. are co-designed.
- Response messages are routed to the AS's in the reverse order

- If during call handling procedure an AS retargets the SIP request by changing the Request URI, subsequent filter analysis in the S-CSCF is stopped and the S-CSCF forwards the request towards the new target without linking-in the other AS' specified by IFC.

Different modes of operations for IMS Application Server (AS) (aka a SIP server)



An AS may either **receive** or **initiate** SIP requests

When it receives an initial SIP request and it does not decide to serve it, the AS may either decide to **remain in the dialogue** initiated by the initial request **or not**. If not, it will only receives answers to the initial request

The following cases are therefore **impossible**:

- The AS gets involved in the dialogue **after** it has started
- The AS **cherry-picks** messages in a dialogue
- An AS involved in a dialogue **drops out** before the end

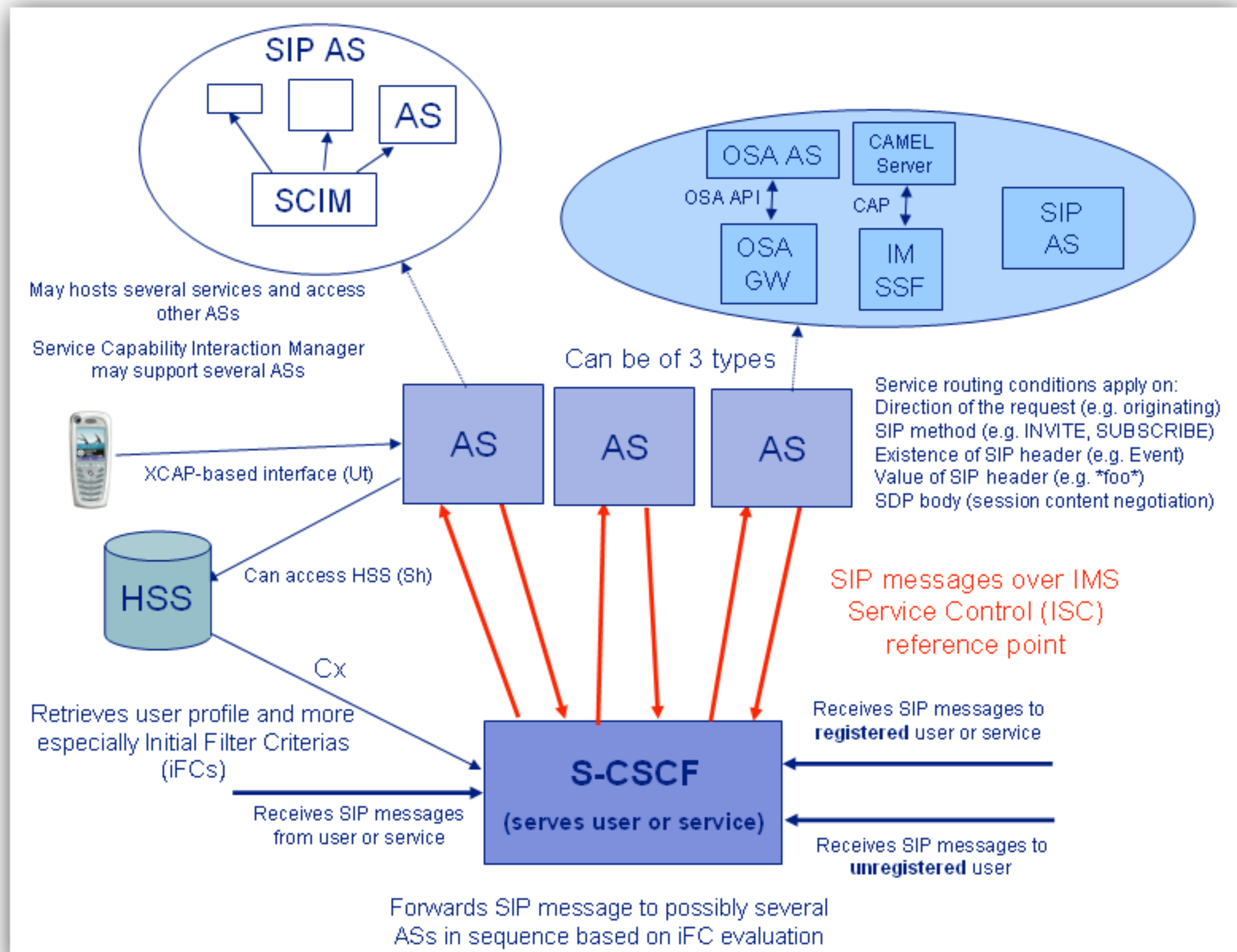
The AS receives **important information** in incoming requests, including whether the user was authenticated or not (see next slide)

The AS may generate requests **on behalf of a user** or **on behalf of a service (public service identity)**

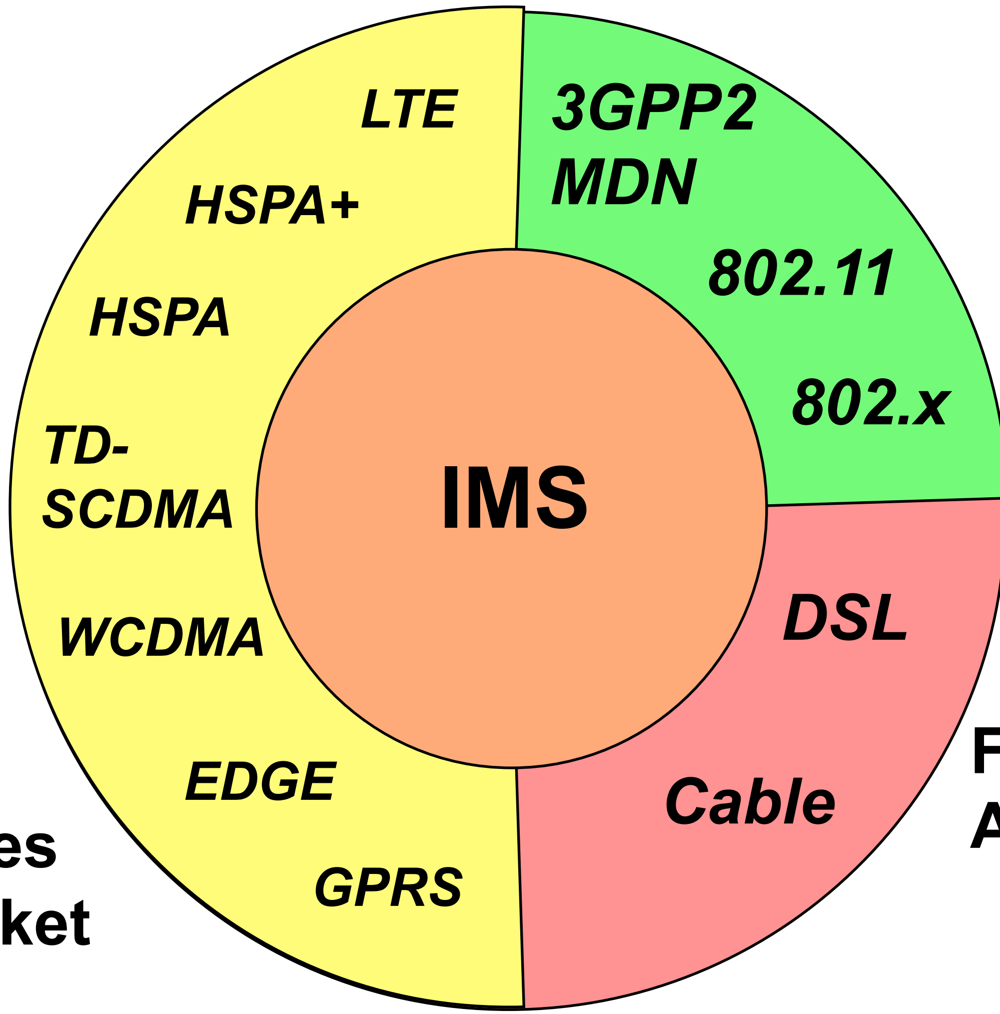
The AS has to support a **specific behavior** when it generates requests, e.g. routing the request to the S-CSCF serving the user on behalf of whom it generates the request, inserting and populating a charging header

Also redirect and registrar…

IMS Service Routing in the S-CSCF using Initial Filtering Criteria in User's Service Profile



IMS is Access Independent (in Theory)

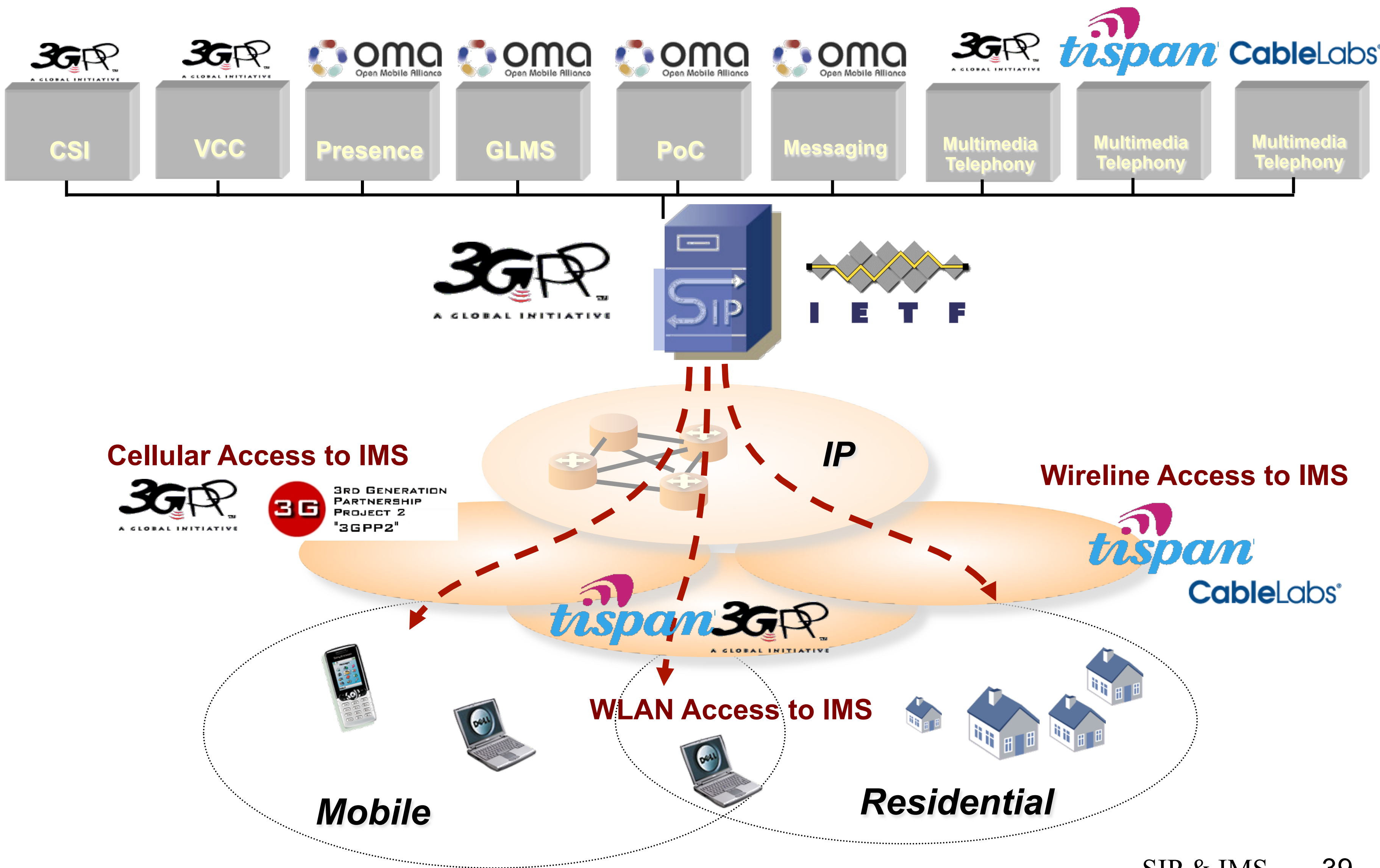


3GPP Defined Radio Accesses For Packet

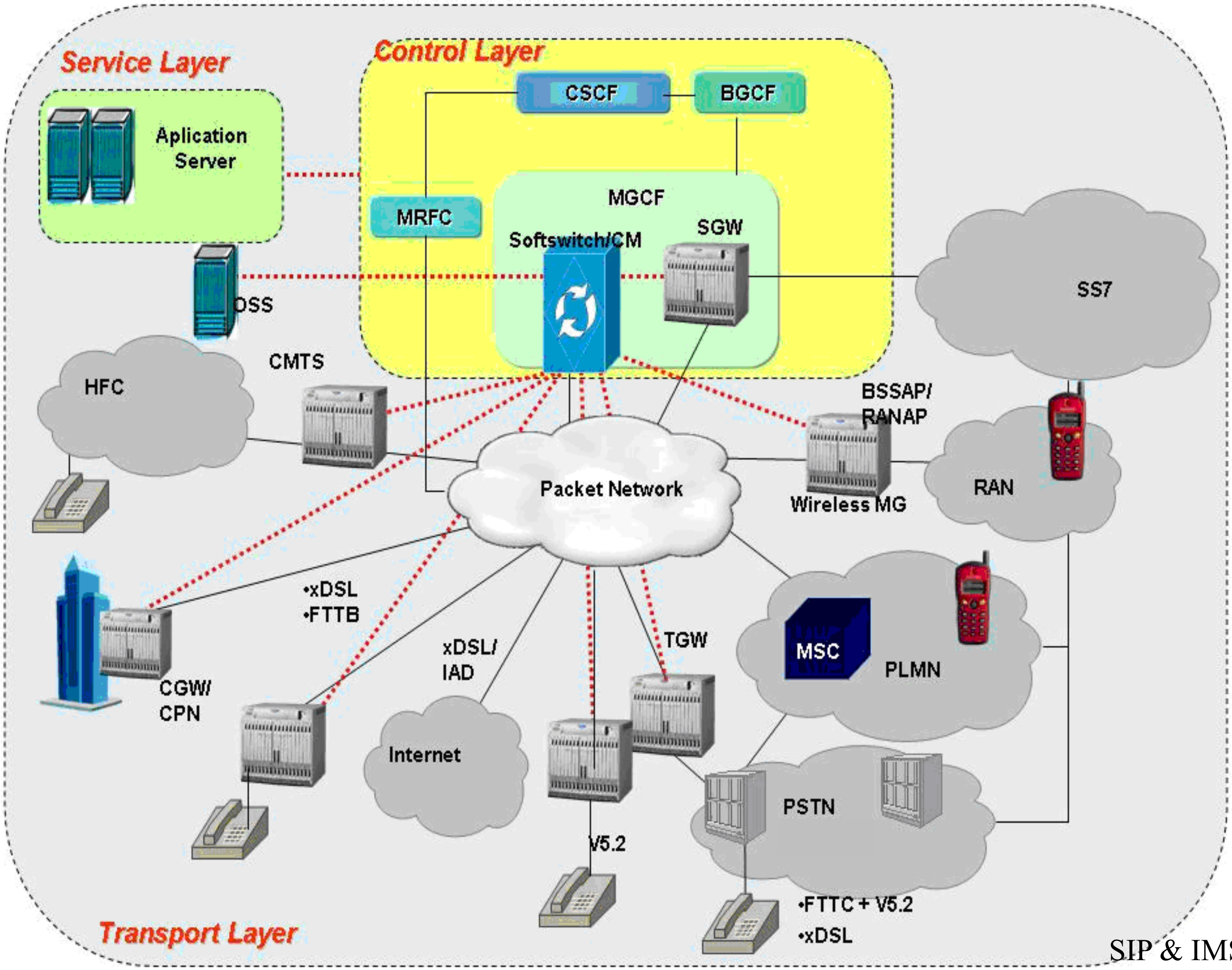
Non-3GPP Defined Radio Accesses

Fixed Accesses

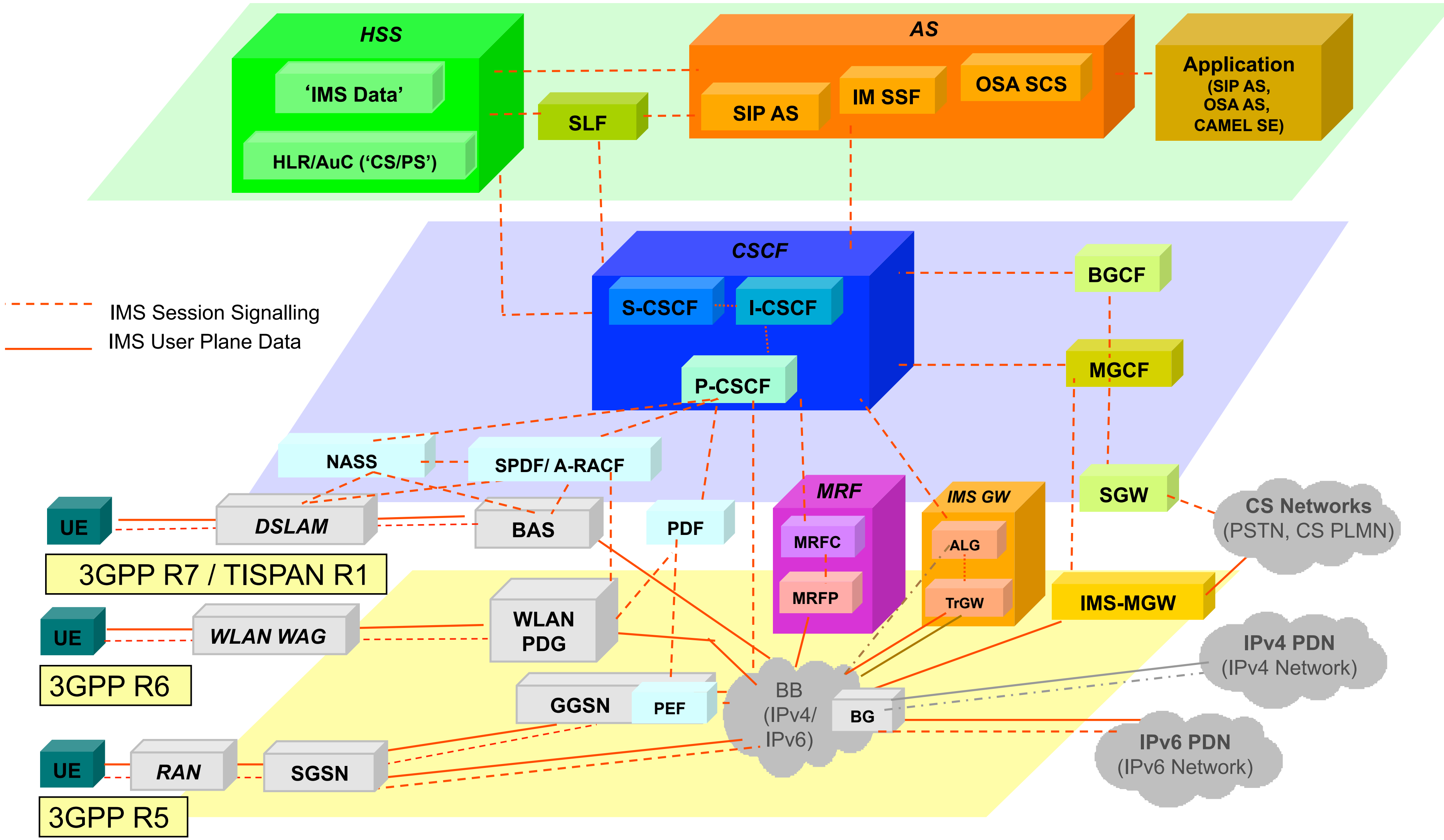
Organizations using IMS



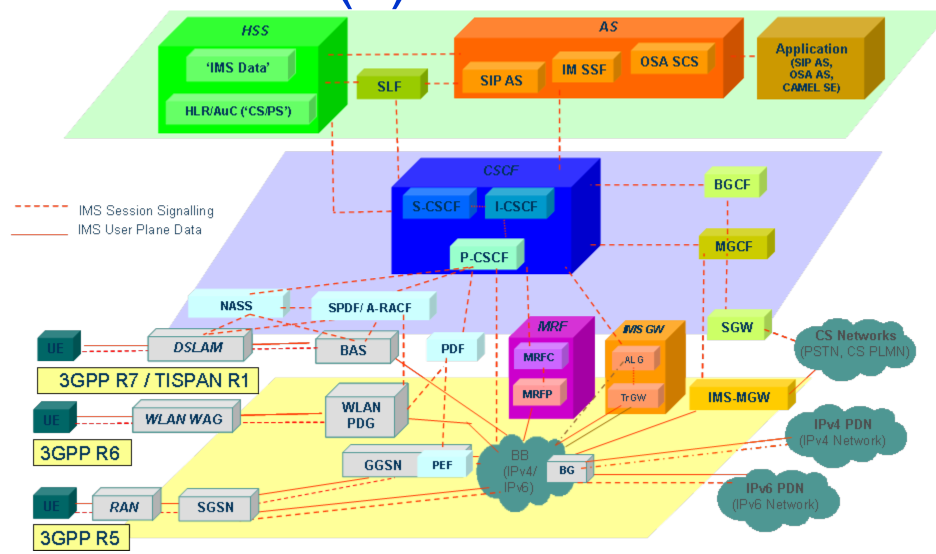
Interworking between VoIP, PSTN and Cellular Services



A more detail IMS Architecture
(incl' Fixed Mobile Convergence w/ TISPAN)



Functional Overview (1)



- CSCF (Call Session Control Function) consists of 3 separate functions: P-CSCF, I-CSCF, S-CSCF

- P-CSCF (Proxy-CSCF):
  - Entry point to IMS from any access network
  - Performs integrity protection
  - Local outbound stateful proxy for all SIP requests/responses, ensuring all signalling is sent via the home network
  - Includes a Policy Decision Function (PDF) that authorizes bearer resources

- I-CSCF (Interrogating-CSCF):
  - First contact point in home network
  - Selects assigned S-CSCF
  - Performs network hiding (THIG)

- S-CSCF (Serving-CSCF):
  - Stateful proxy that provides session control
  - Performs subscriber authentication
  - Acts as SIP registrar
  - Invokes the AS' (Application Servers) based on IFC (Initial Filter Criteria)

- **SLF (Subscriber Location Function):**
  - Look-up function used in networks where multiples HSS' exist

- HSS (Home Subscriber Server):
  - IMS subscriber records and service profile
  - IMS authentication data

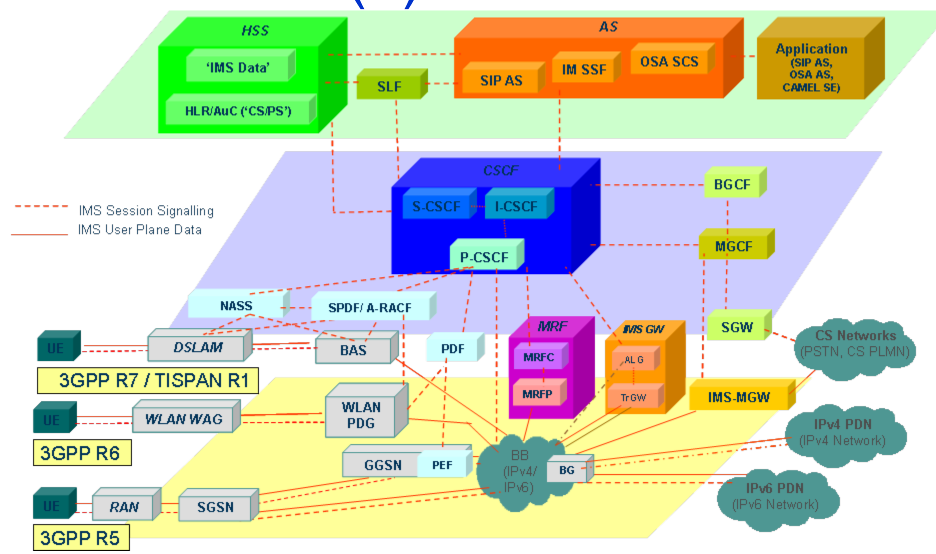
- MRF (Media Resource Function) consists of 2 separate functions: MRFC, MRFP

- MRFC (Media Resource Function Controller):
  - Controls media resources in MRFP
  - Acts as SIP B2BUA

- MRFP (Media Resource Function Processor):
  - Media stream processing (transcoding etc.)
  - Multimedia announcements
  - Incoming streams mixing

Functional Overview (2)

- SIP AS (Application Server):
  - Hosts IMS native applications
- **IM SSF** (IP Multimedia Switching Service Function):
  - Provides interworking with CAMEL, ANSI-41, INAP or TCAP services
- **OSA SCS** (Open Service Architecture Service Capability Server):
  - Provides interworking with OSA services
- BGCF (Breakout Gateway Control Function):
  - Selects the network in which PSTN breakout is to occur and within that network selects the MGCF
- MGCF (Media Gateway Control Function):
  - Controls media channels in IMS MGW
  - Performs conversion between ISUP/TCAP and IMS call control protocols
- IMS MGW (IMS Media Gateway):
  - Terminates bearer channels from CS networks and PS media streams
  - Owns/handles resources (echo cancellers, codes, etc.)



- SGW (Signaling Gateway):
  - Performs conversion at transport level (SCCP, SCTP)
- **PDF/SPDF** (Policy Decision Function / Serving Policy Decision Function)
- **PEF**: Policy Enforcement Function
- **A-RACF** (Access - Resource and Admission Control Function):
- **NASS** (Network Attachment Subsystem):
- **DSLAM** (Digital Subscriber Line Access Multiplexer):

Sample Services made possible by 3GPP IMS/ SIP

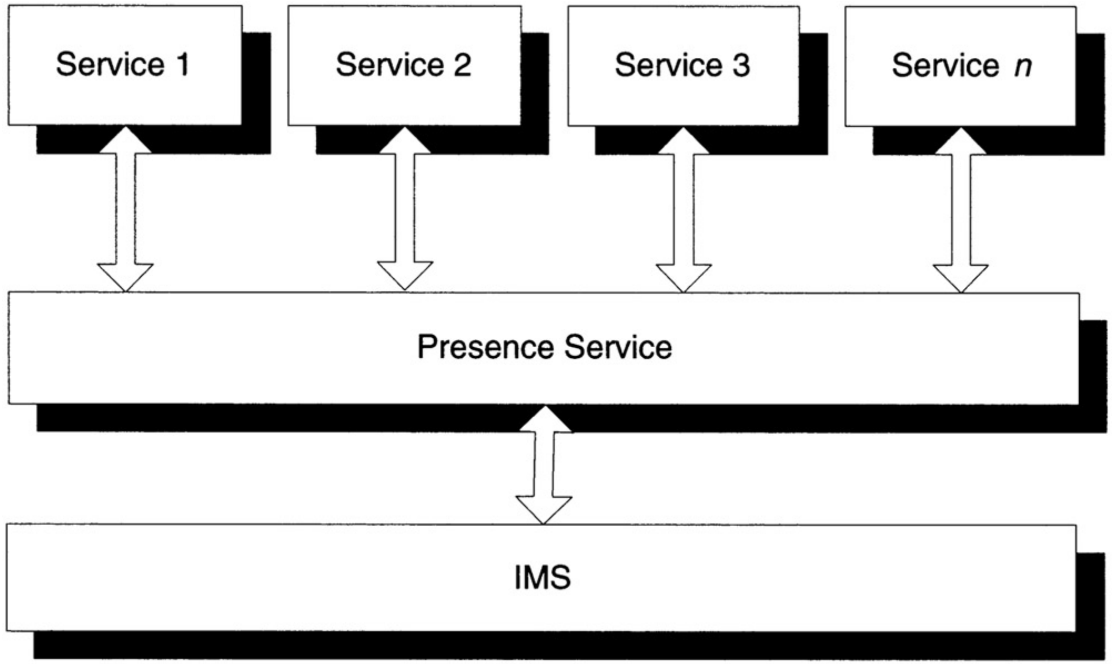
- Presence Service
- Group Management
- Push Service
- Instant Messaging
- Multi-party Multimedia Conference
- Location-based service
- E2E QoS signaling, policy control and enforcement
- Voice Call Continuity service (VCC)

IMS Presence Service

- User defined visibility to others
  - e.g. reachable for everybody by any communication means when online.
  - Except when in a meeting. Then only reachable by email. Unless it is the boss, then also available by phone
- User can find out presence of others
- Other services can use this service
  – Push services, push-to-talk,…
- Supported via…
  - Presence Server (called Presence Agent in IETF terminology)
    - A SIP Application Server
    - Stores all presence information
    - Provides information on user presence obtained from UE or network
  - Watcher Application / Proxy
    - Request specific presence information from Presence Agent
      - Upon-request (“pull“)
      - By subscription (“push“) (e.g. alert when user becomes available)
  - Standardized Format for presence information, access rules

More details later on how the IETF SIP-based SIMPLE framework can support this service

Presence as the Foundation of ALL IMS services



- Presence services can help NOT only end-users BUT ALSO other services in IMS, e.g.
  - An Voice-mail server can send a Notification message to a user when he/she returns to online status
  - A video server can select the best o/p format and coding-rate depending on where and how the viewer's current device is connected to the network

IMS Group Management

- Setting up and maintaining user groups
  - Uses Presence Service
- Supporting service for other services
  - Multiparty conferencing
  - Push-to-talk, etc

IMS Messaging

- SIP-based messaging
- Instant messaging, “Chat room“, and deferred messaging (equivalent to MMS)
- Interwork with Presence Service to determine whether addressee is available

IMS Multiparty Multimedia Conference service

- Utilizing MRF (Media Resource Function)
- Supported by Group Management Service

IMS Location-based services in IMS

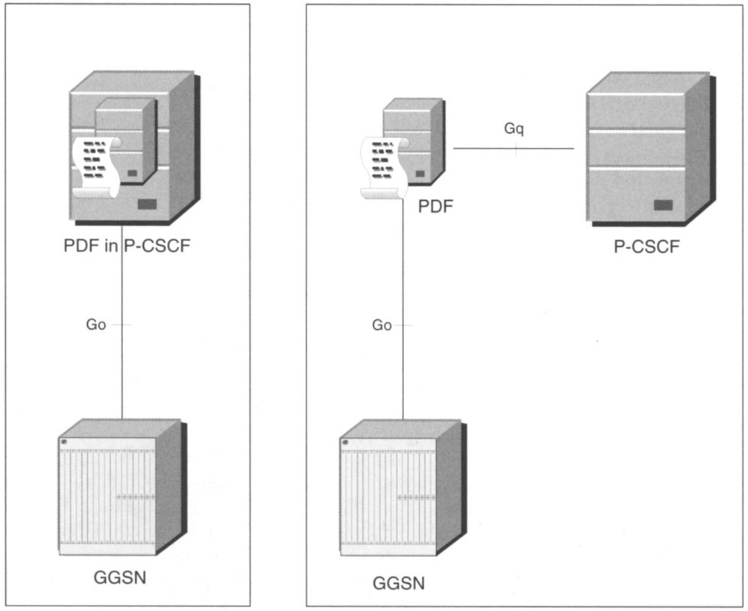
- UE indicates it wishes to use local service. S-CSCF routes request back to visited network
- Mechanism for UE to retrieve / receive information about locally available services

End-to-End QoS Signaling, Policy Control and Enforcement in 3GPP IMS using SIP

End-to-end QoS Signaling and Call Admission Control (CAC) in 3GPP IMS

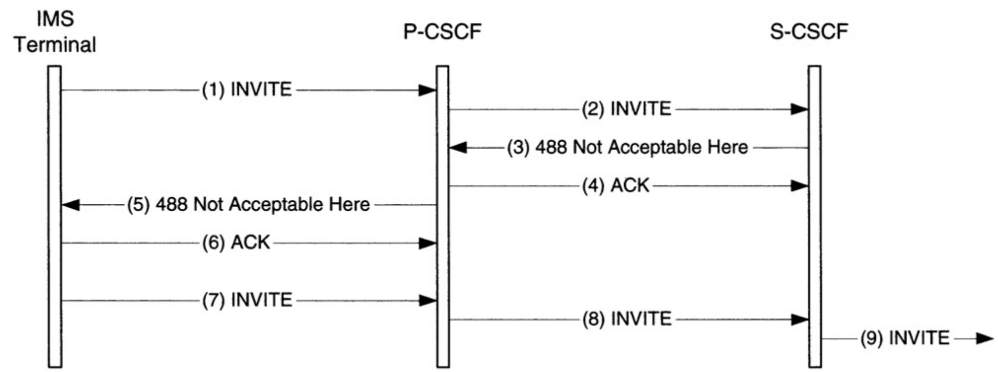
- IMS provides correlation and control mechanism using the Policy Decision Function (PDF) which
  - Act as Policy Decision Point (PDP)
    - The GGSN is the corresponding Policy Enforcement Point (PEP)
  - Authorize and Control resource usage for each Bearer (i.e. GPRS/UMTS PDP-context) to:
    - Prevent misuse of Network QoS/ Theft of Service
    - Allow throttling of resource consumption
  - Exchange Charging Correlation identifiers with GGSN
    - Enable correlation of charging info generated in the PS-domain (SGSN, GGSN) and in the IMS (e.g. CSCF, AS)

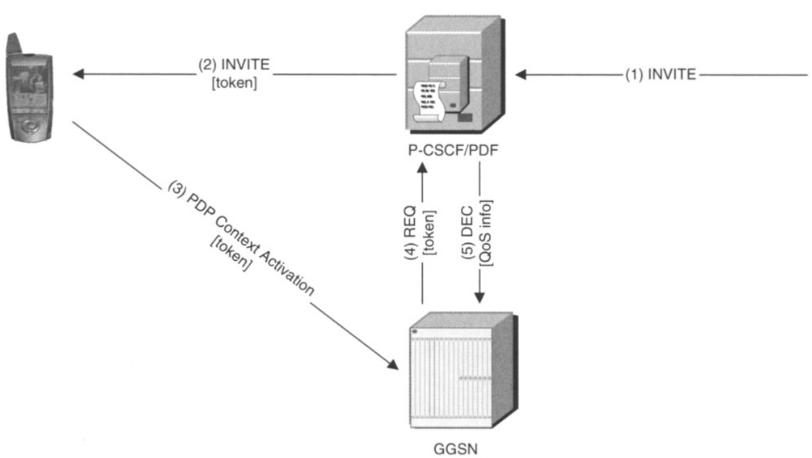
End-to-end QoS signaling/ Call Admission Control (CAC) with IMS (cont'd)

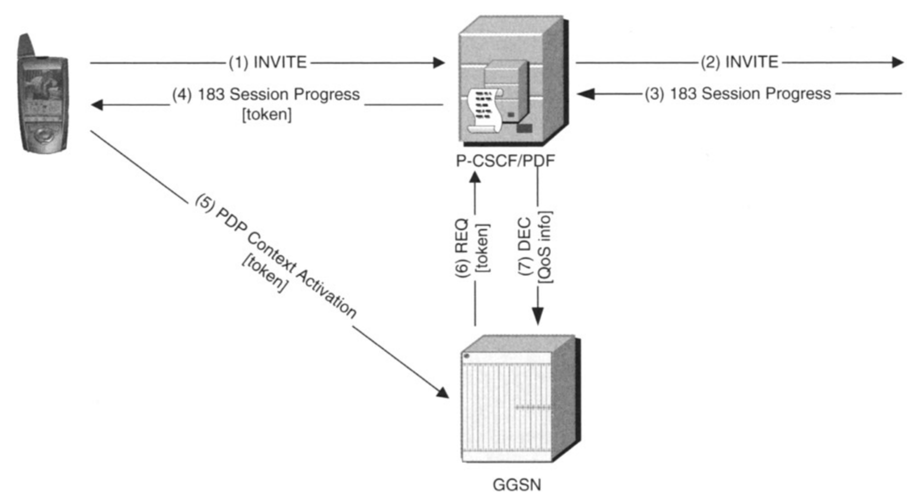


3GPP Release 5

3GPP Release 6

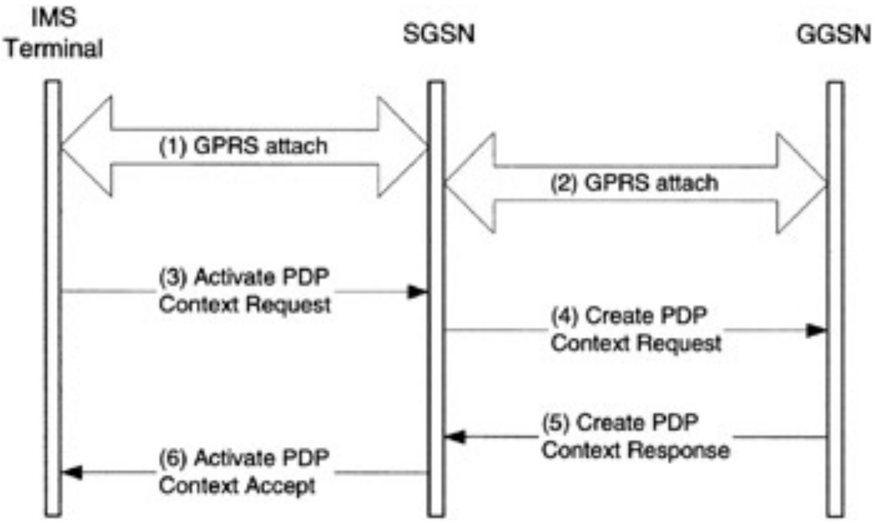


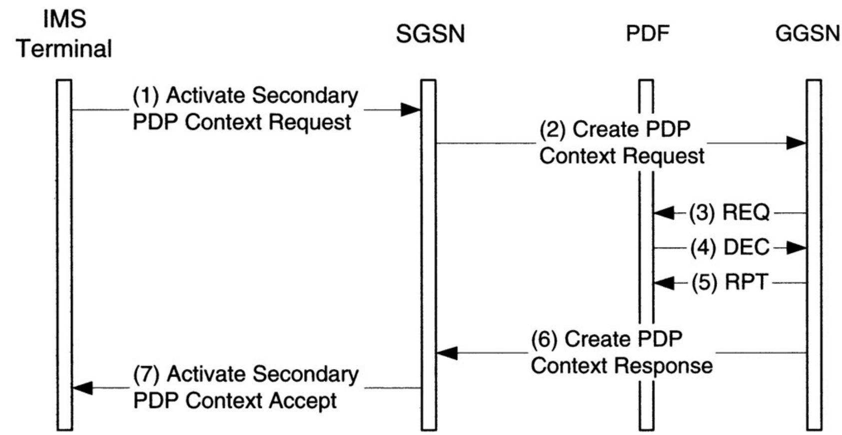


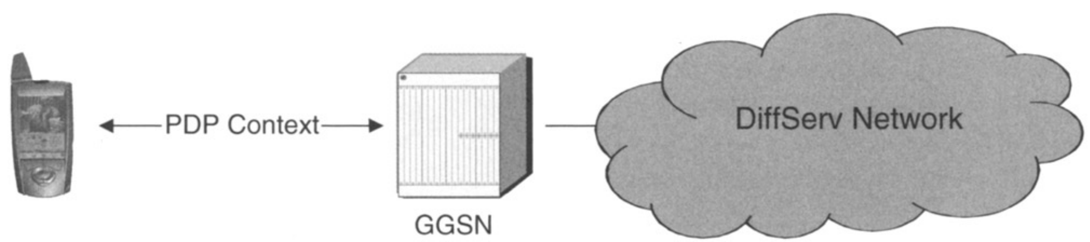


SIP Procedures for Media Authorization

QoS Request and Authorization w/ IMS







IMS QoS Support

- IMS provides correlation and control mechanism using the Policy Decision Function (PDF) which
  - Act as Policy Decision Point (PDP)
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  - Authorize and Control resource usage for each Bearer (i.e. GPRS/UMTS PDP-context) to:
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    - Allow throttling of resource consumption
  - Exchange Charging Correlation identifiers with GGSN
    - Enable correlation of charging info generated in the PS-domain (SGSN, GGSN) and in the IMS (e.g. CSCF, AS)

Backup Slides

3GPP Voice Call Continuity via IMS support

What is 3GPP VCC (TS 23.206, 24.206) ?

- VCC supports Real-time Voice Call continuity when users move between 2G-GSM/3G-UMTS Circuit-Switched (CS) Domain & IP Connectivity Access Network (IP CAN), e.g. WLAN interworking, with home network IMS functionality
  - In theory, the scheme should also work for VCC between 2G/3G CS domain and any IMS-based VoIP calls
  - In reality, previous 3GPP specs do not support simultaneous transmit/receive of 2G CS and 3G-UMTS PS radios
  - VCC between 2G-CS and IMS accessed via 3G-UMTS/PS domain IP-CAN becomes out of scope

  In short:
  - Main application in 2G/3G CS voice interworking with WLAN
  - VCC of 2G-CS and VoIP-using 3G CDMA PS service is not supported

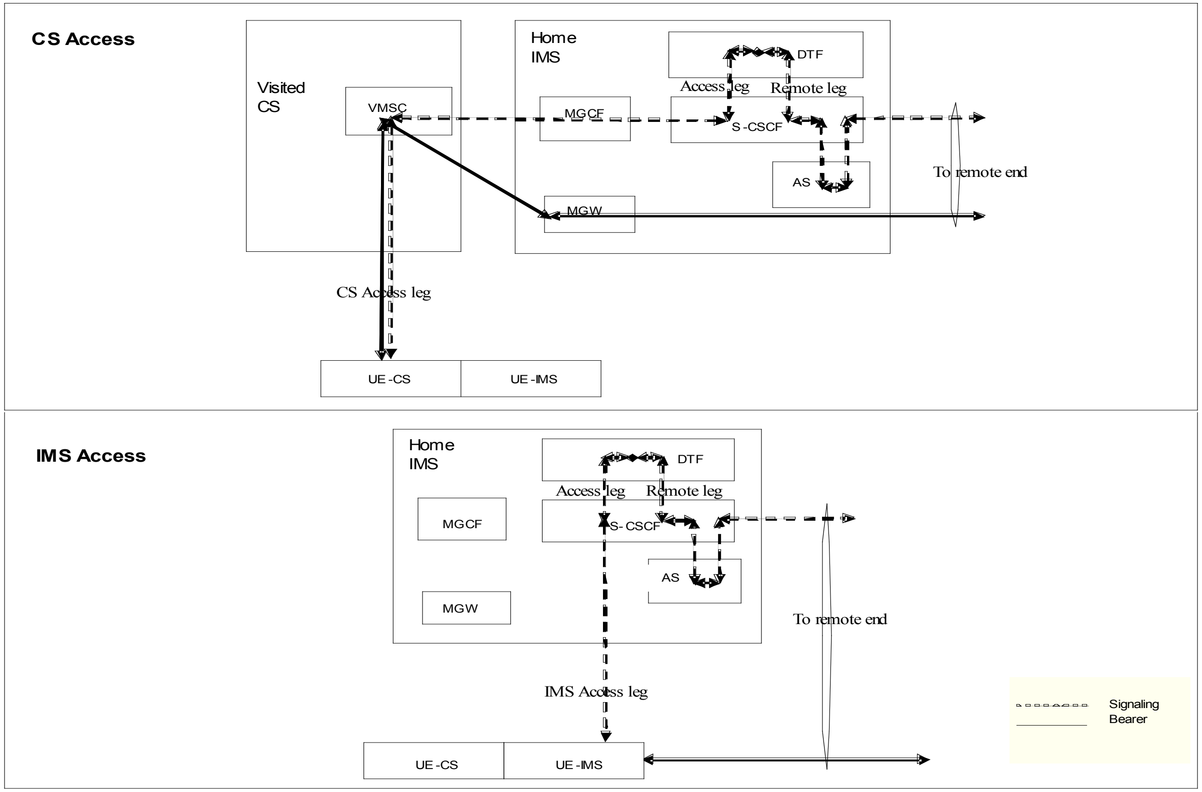
- VCC of Emergency Call not yet supported

Mechanisms for 3GPP VCC

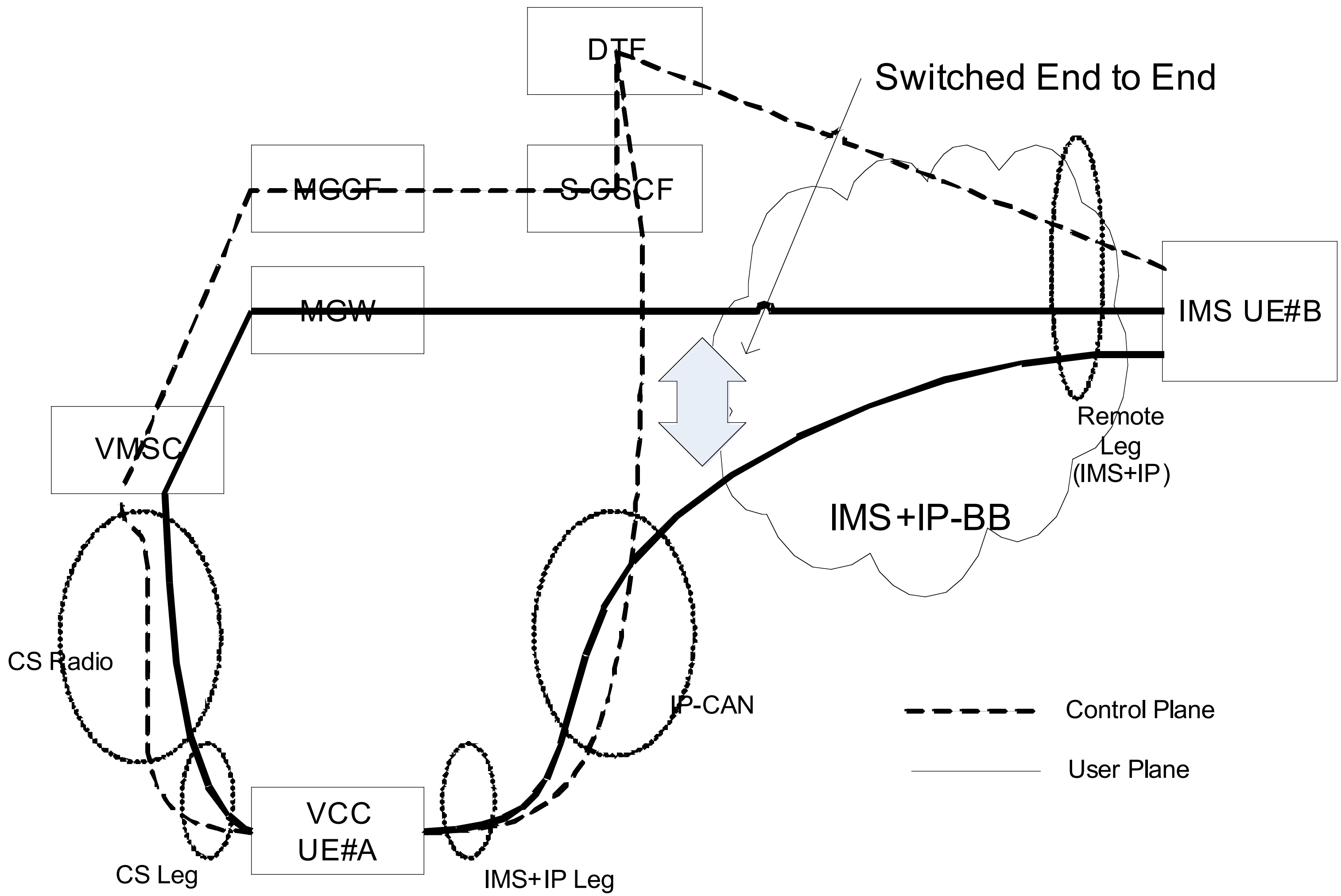
- The call of a VCC UE is anchored at VCC application (a SIP AS) in the Home IMS
  - Anchor = Insertion of Domain Transfer Function in signaling path of the Voice call
  - For a CS originating call, standard CS domain techniques are used to reroute calls to IMS at call establishment
- 3pcc (3rd party Call Control) function is employed at the VCC application to provide mobility b/w IMS and CS domain
  - Use the Back-to-Back SIP User Agent mode of operations of an AS
- VCC domain transfer is initiated by the VCC UE and executed as an VCC application in the home IMS
- Other key functional blocks inside the VCC application include:
  - Domain Selection Function (DSF)
  - Domain Transfer Function (DTF) and
  - CS Adaptation Function (CSAF)

3rd Party Call Control (3pcc) at home IMS

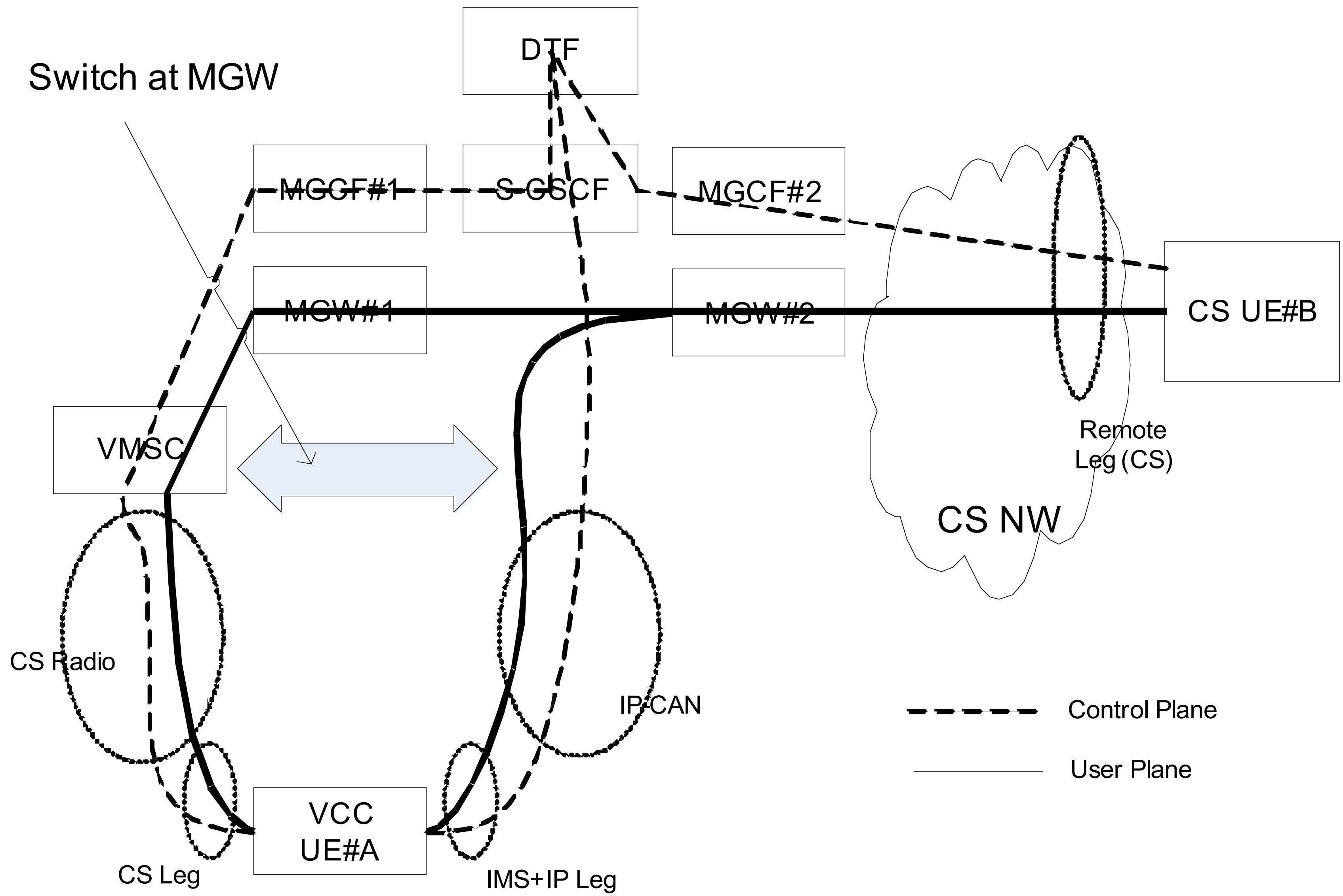
- For VCC UE originating calls:
  - Follow IMS/iFC standard mechanisms to forward call-setup to home DTF where DTF terminates the access-leg and then uses 3rd Party Call Control (3pcc) to initiate call to remote party
- For VCC UE terminating calls:
  - Incoming call setup request (remote leg) is first terminated by the DTF, which in turn establishes an access-leg towards the VCC UE



User Plane Path b/w VCC UE and IMS UE



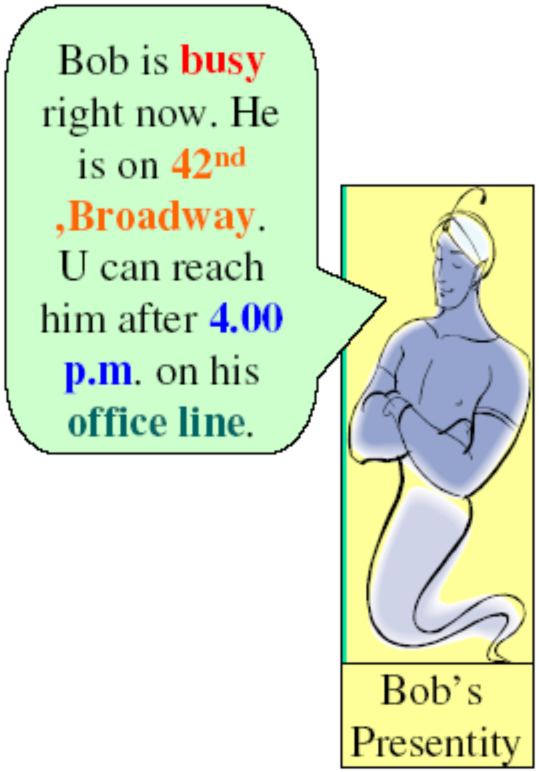
User Plane Path b/w VCC UE and CS UE/PSTN



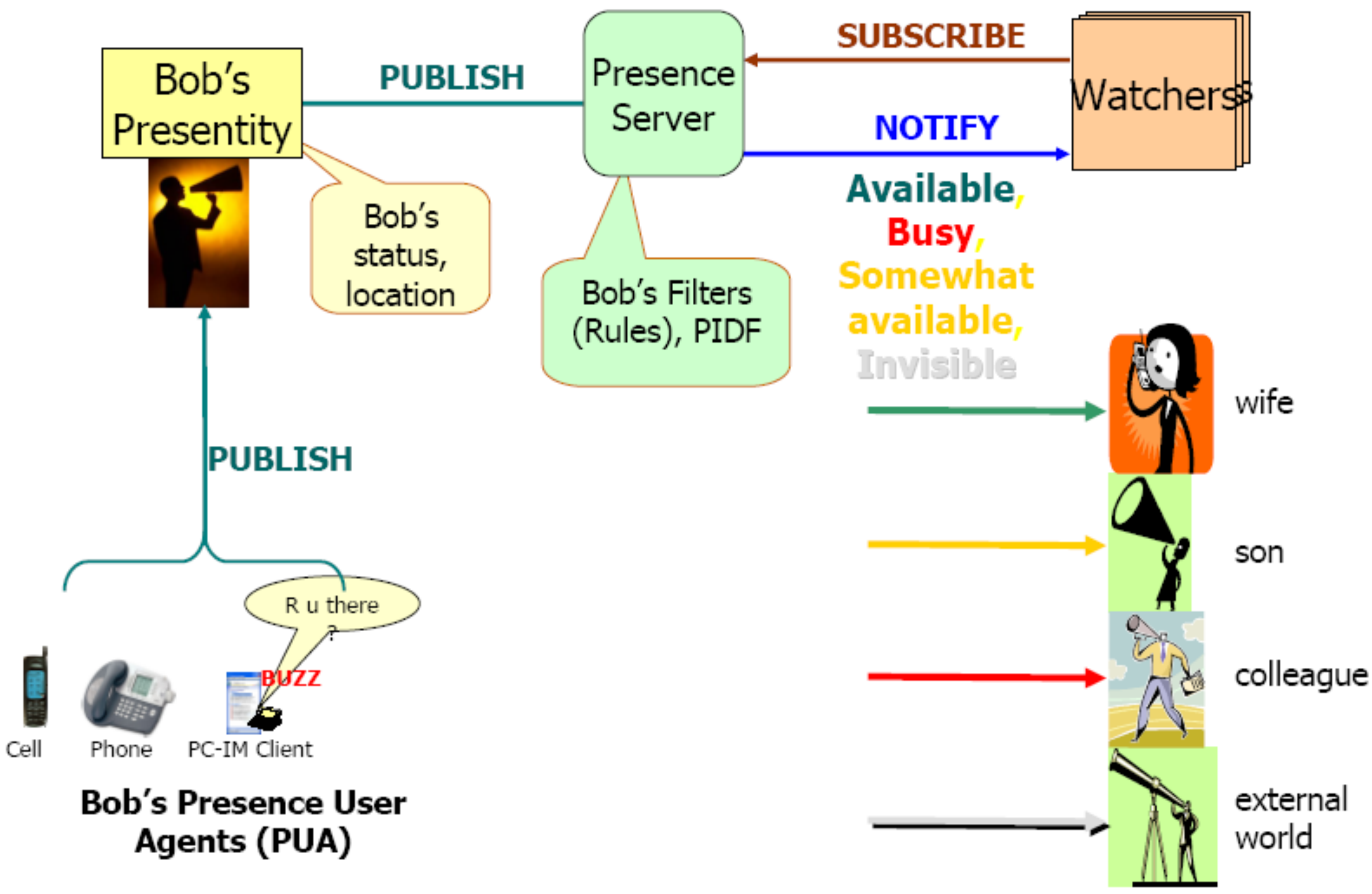
Presence Services via the IETF SIMPLE Framework

Presence System Overview

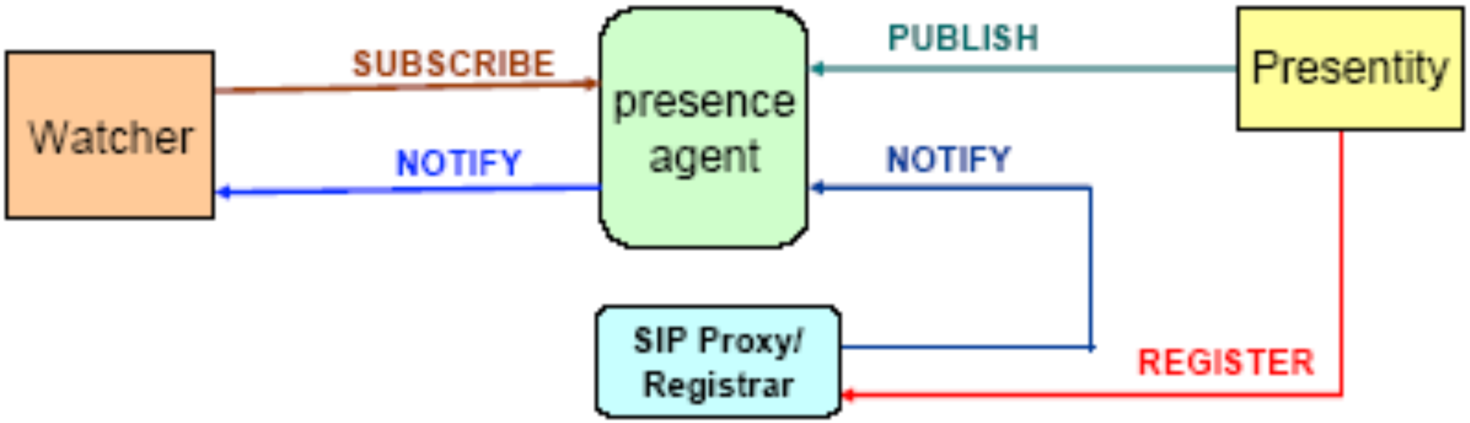
- Presence:
  - Ability and Willingness to communicate
  - Rules about HOW and WHAT part of presence info can be accessed
  - More detailed info (aka Rich Presence) can include:
    - Location, preferred communication mode, current mood/activity
- Presentity
  - Represent a user or a group of users of a program
  - Source of presence information
- Watcher
  - Requester of presence info about a presentity



Presentity and Watchers

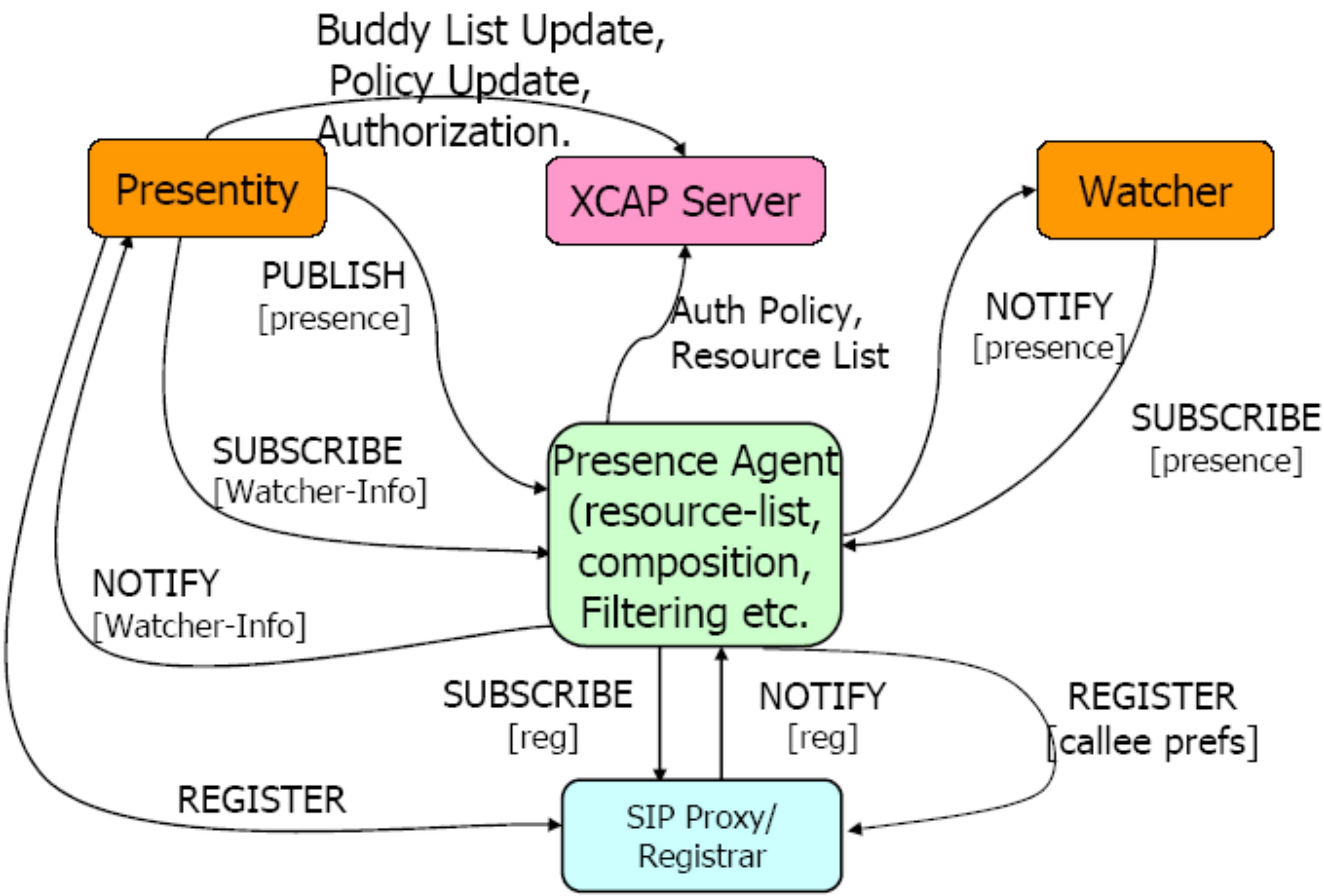


The IETF Presence Framework
SIMPLE: SIP for Instant Messaging and Presence Leveraging Extensions

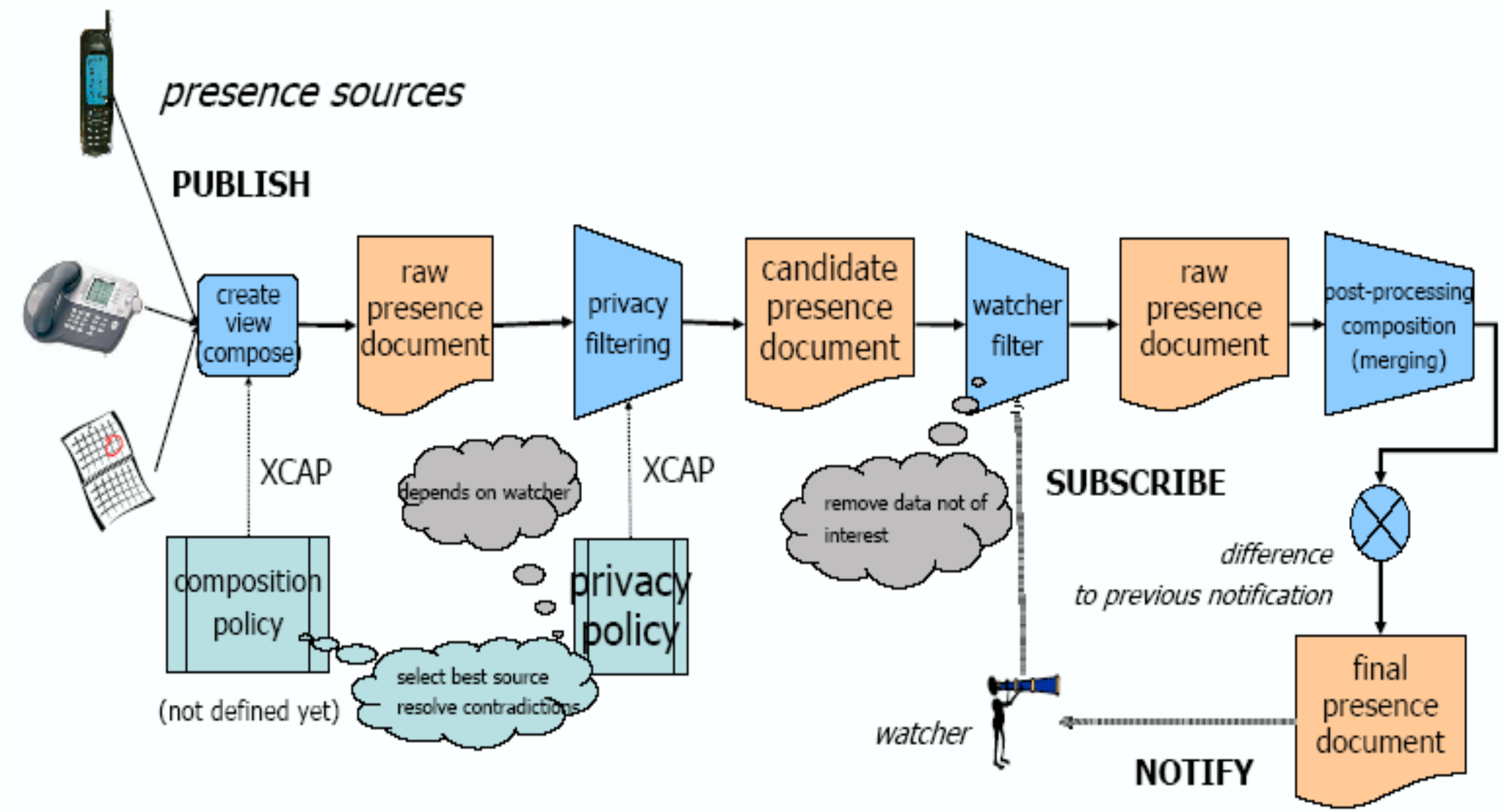


- Basic Presence Components:
  - Presence Agent (aka Presence Server)
    - Provides information on user presence obtained from UE or network
    - A SIP Application Server
    - Stores all presence information
  - Watcher Application / Proxy
    - Request specific presence information from Presence Server
      - Upon-request (“pull“)
      - By subscription (“push“) (e.g. alert when user becomes available)
- Realized based on the SIP event package

Presence Configuration



Presence Data Model and Data Processing



Presence Data Formats and Extensions

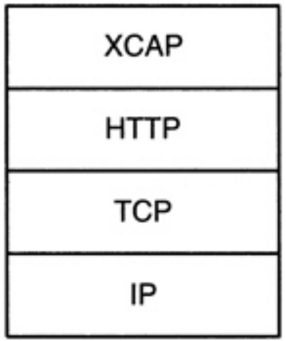
- PIDF (Presence Information Data Format)
  - XML format
  - Easily extensible

- RPID (Rich Presence Information Data)
  - Extension of PIDF to include detail info about the presentity, e.g.
    - Current activity
    - Mood

- Other extensions of PIDF include:
  - Time Presence
  - Partial Presence: PIDF-diff
    - only send differences to optimize bandwidth usage
  - Location (called PIDF-LO in IETF Geopriv Working Group)
  - User Agent Capabilities (characteristics and preferences)

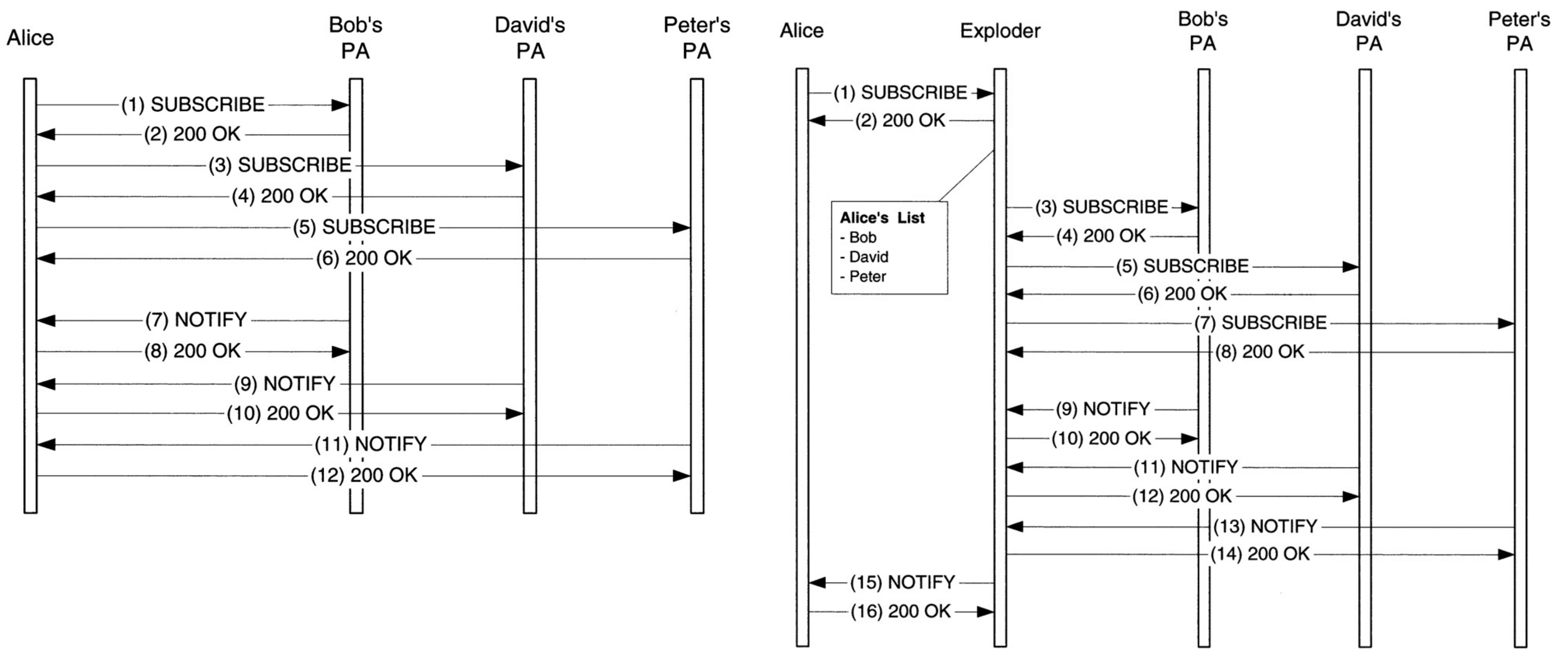
Additional Components

- XCAP Server
  - To store and manage authorization/privacy configuration rules e.g:
    - Buddy List
    - Presence Rules
    - Resource Lists
  - Data stored in XML format
    - Different List components correspond to elements of different XML Namespace
    - Can be accessed/updated via HTTP get/put/delete methods
  - The key idea of XCAP is to use HTTP methods to read/update the specific elements defined in the XML namespace



Additional Components (cont'd)

- Event Registration Package
- Watcher Info Package
- Resource List package

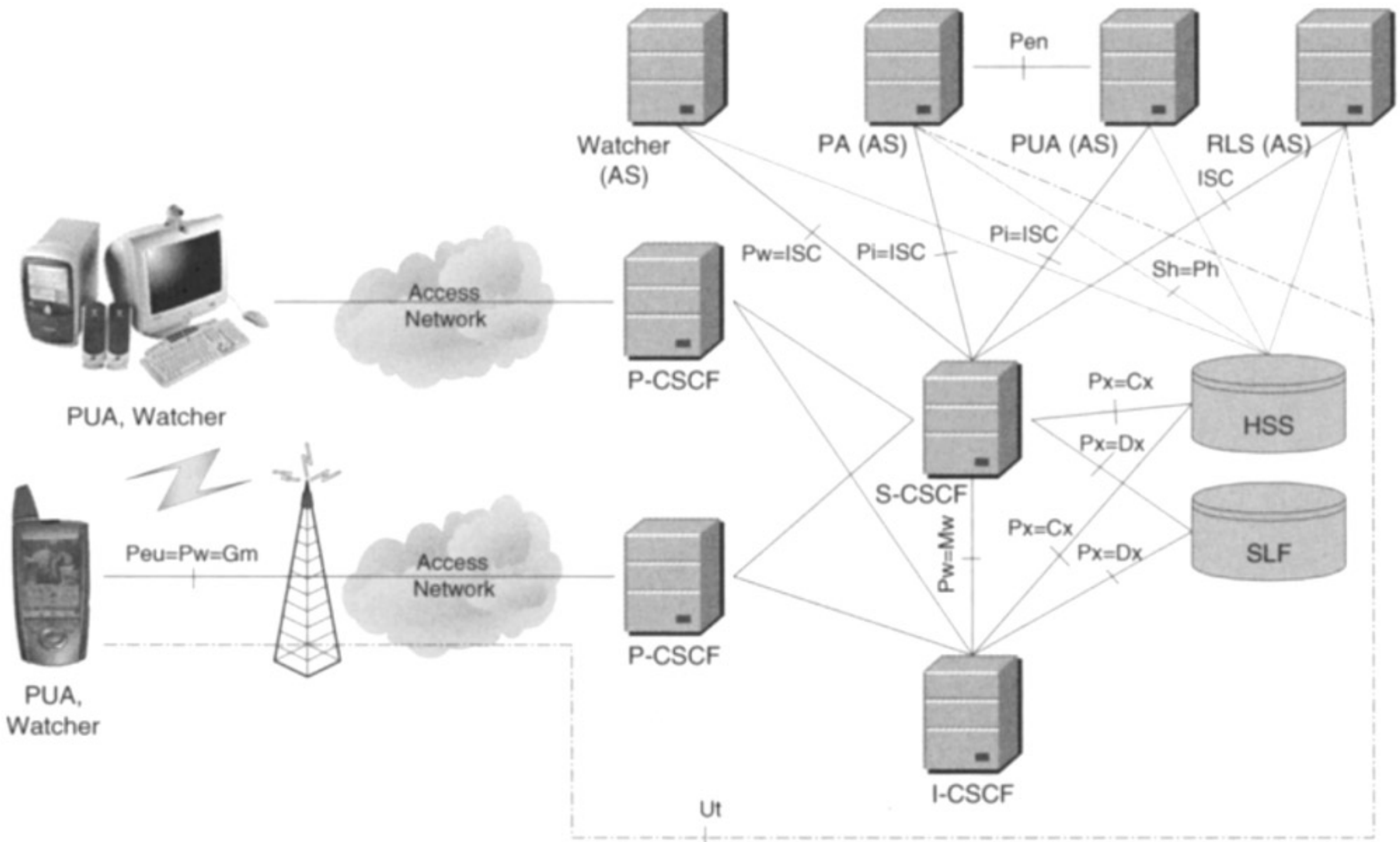


Without Vs. With Exploder and Resource List

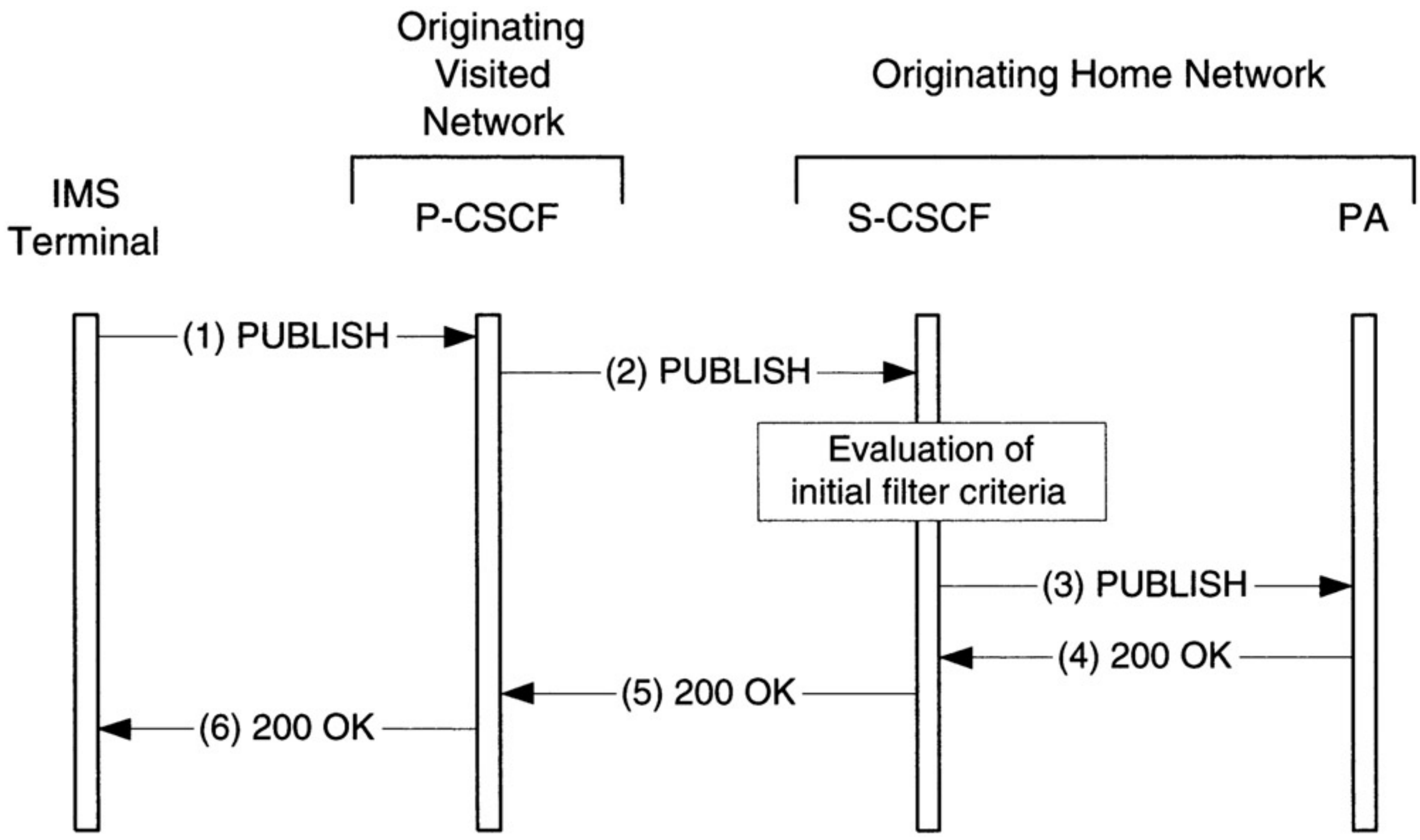
More IETF SIMPLE References

- Jonathan Rosenberg, “SIMPLE made simple: An Overview of the IETF Specifications for Instant Messaging and Presence using the Session Initiation Protocol (SIP)”, IETF Draft draft-ietf-simple-simple-01, Nov. 2007.

IMS Presence Architecture (based on IETF SIMPLE)



Publish Presence Info with IMS

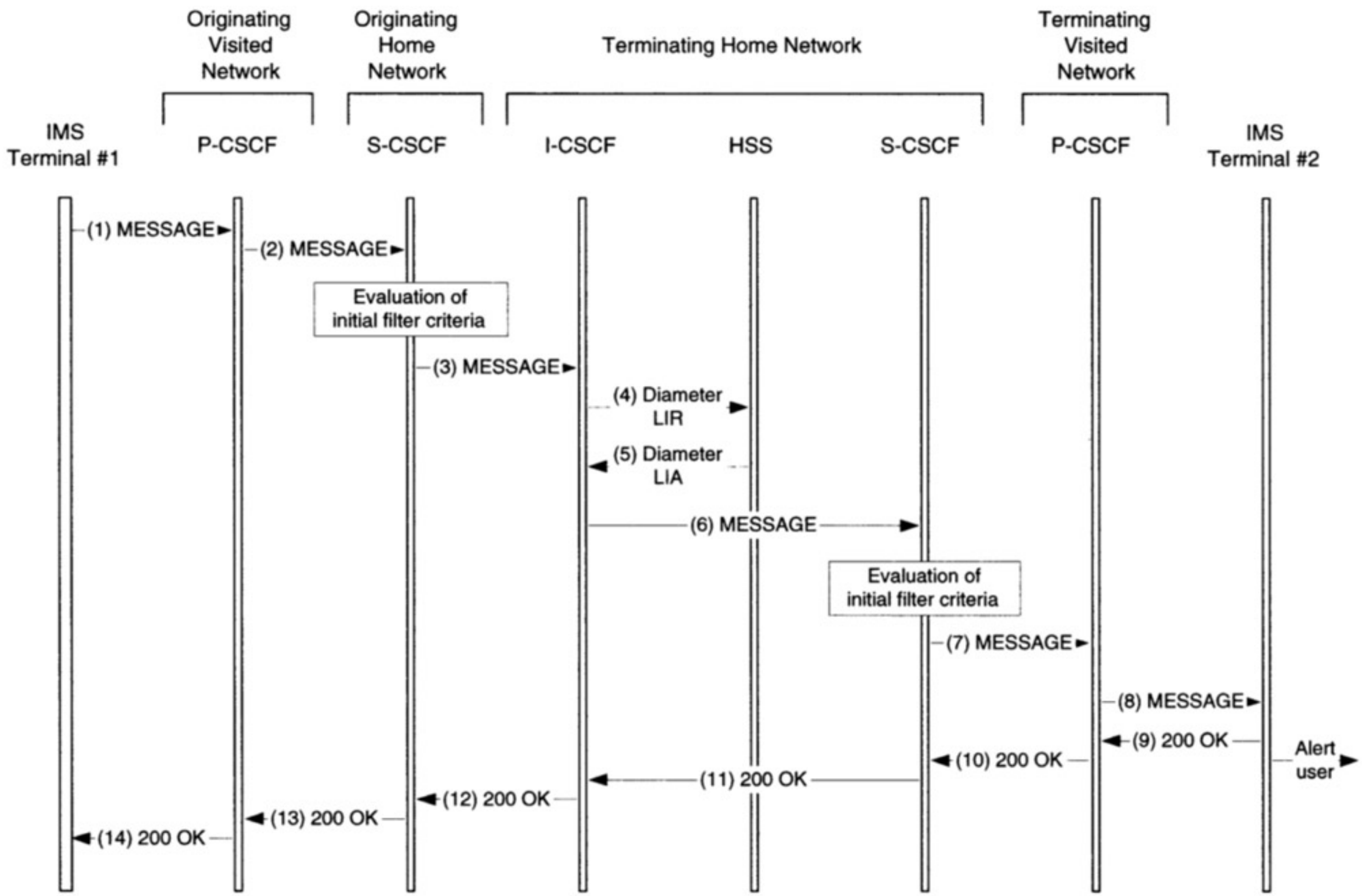


Instant Messaging Support via SIMPLE

Two modes of Instant Messaging operations:

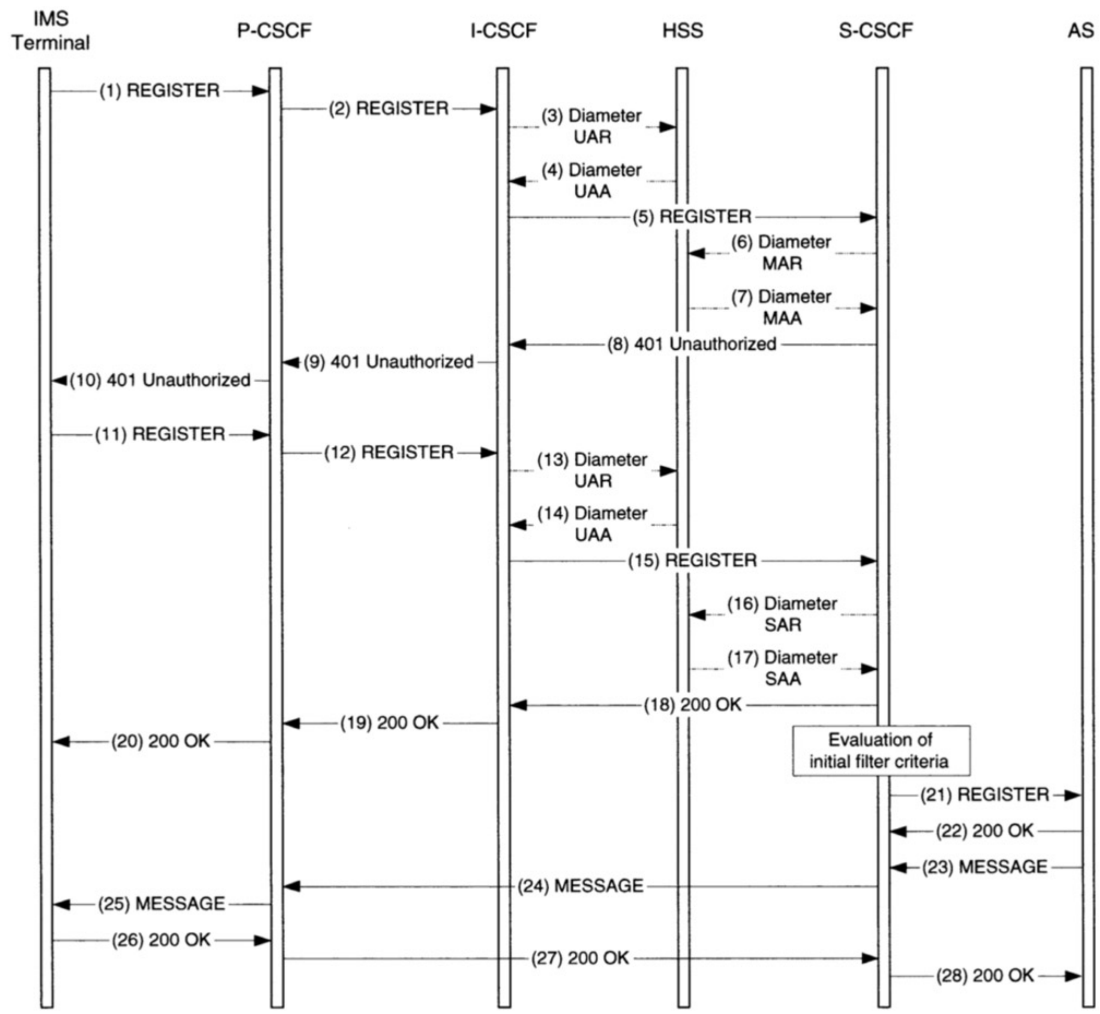
- (Two-way) Pager mode
  - Define a new method in SIP called “MESSAGE”
    - Use MIME to carry encapsulate different types of payload
    - Runs in Control Plane
      - Instant message content will route via SIP servers/proxies
  - No guarantee of congestion control in every SIP-proxy leg
    - Need to impose max. limit on Instant message size (MTU – 200 bytes)
    - Can circumvent limit via SIP content indirection mechanism
      - Inserting HTTP URI in the SIP message for download
- Session mode
  - Use SIP as the session control protocol to setup additional text-based communication session
    - Use Message Session Relay Protocol (MSRP)
    - May incur longer setup latency
  - MSRP runs in media plane, by-passing intermediate SIP proxies
  - Better support for NAT/Firewall Traversal
  - Can use different transport protocols
    - End-to-end Congestion control possible
      - No limit on Instant Message size

Instant Messaging with SIMPLE in IMS



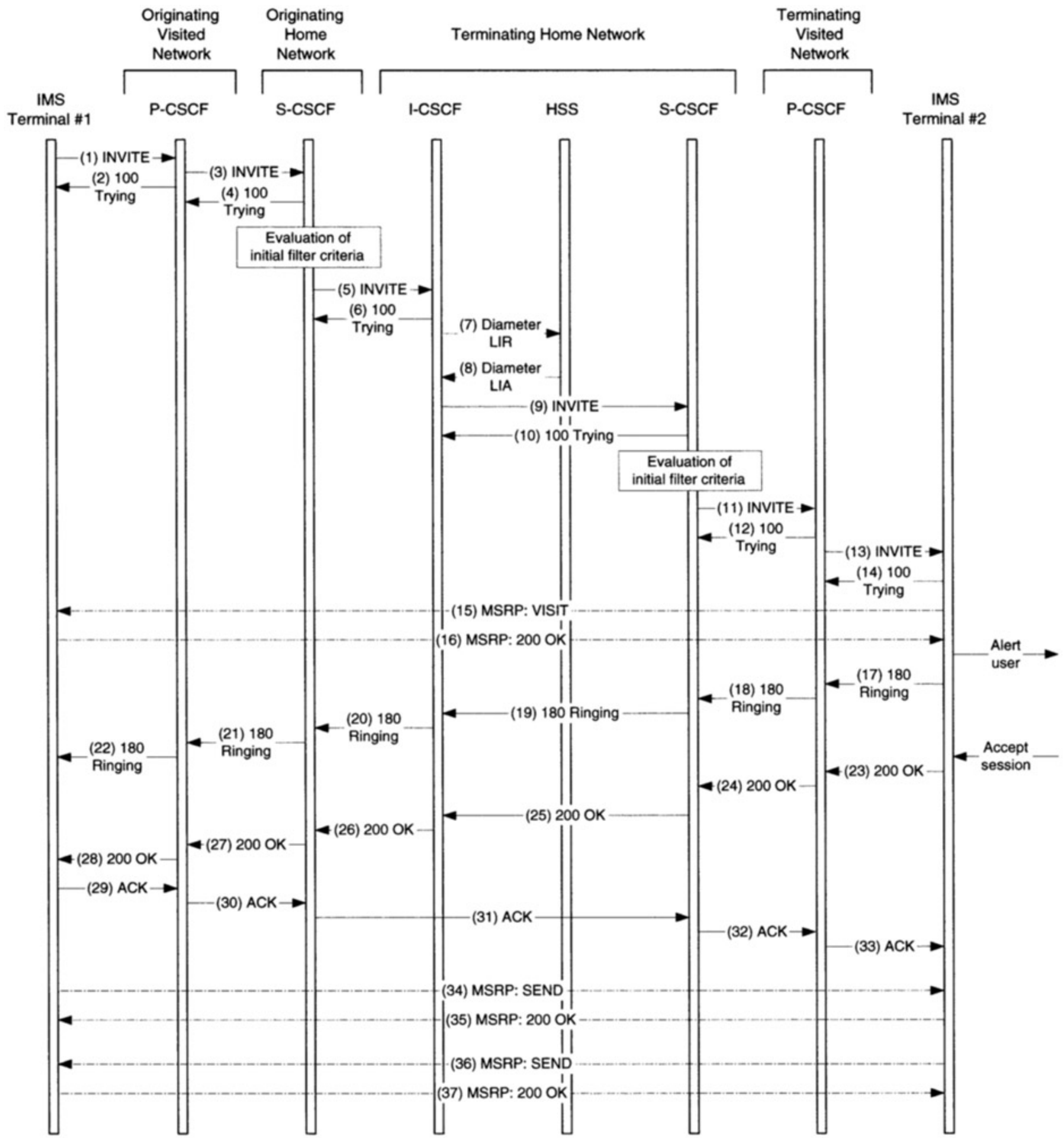
Two-way Pager-mode of Instant Messaging with an end-user

Instant Messaging with SIMPLE in IMS (cont'd)



An IMS Service provided by Pager-mode of Instant Messaging with AS

Instant Messaging with SIMPLE in IMS



Session-based Instant Messaging using MSRP

IETF Protocol to support Location-based Services

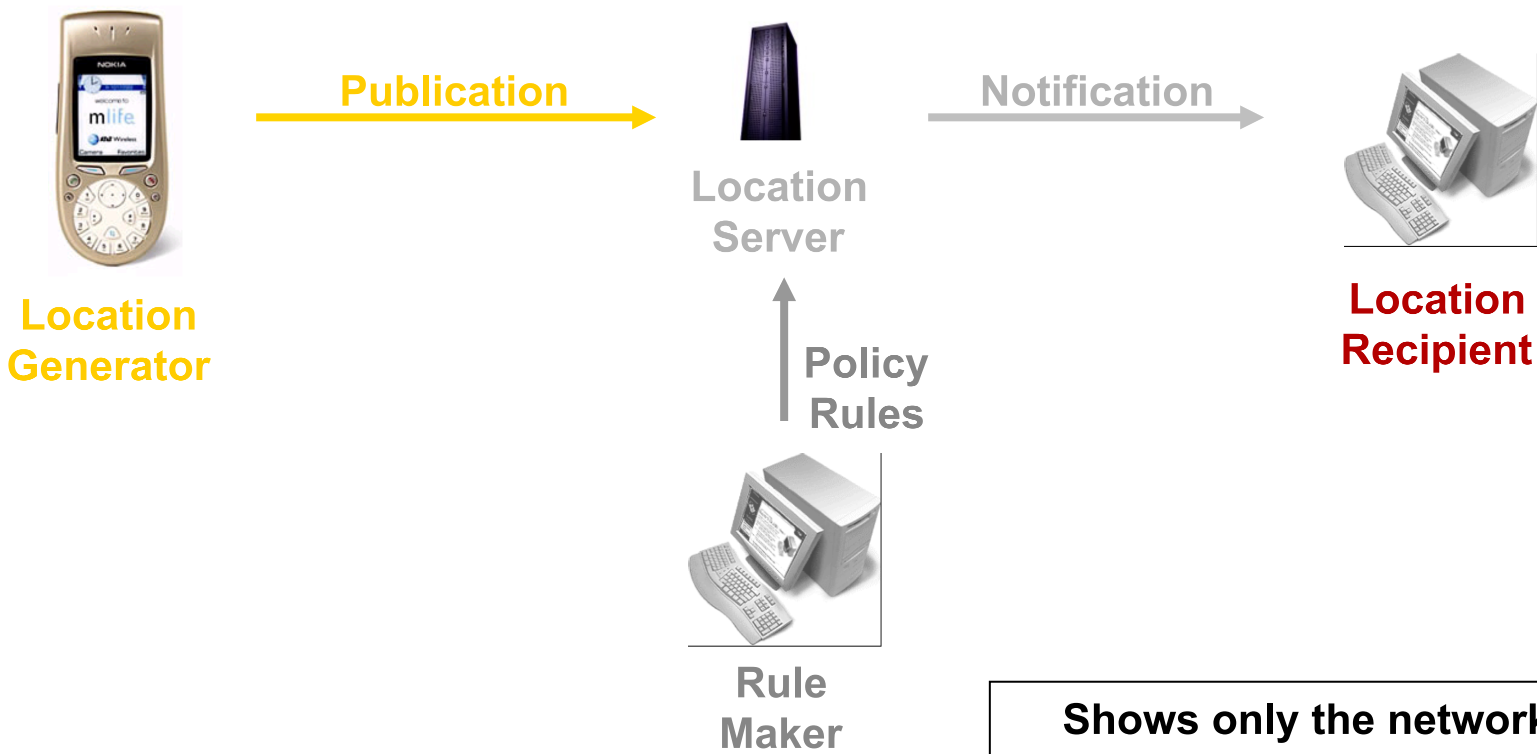
The GEOPRIV IETF Working Group

- First BoF on Spatial Location held at 48th IETF (July 2000)
  - Concerns that privacy was not sufficiently addressed
- GEOPRIV WG formed, met for the first time at 50th IETF (August 2001)
  - Strong user privacy mandate in WG charter
- Scope is exclusively protecting the distribution of location information over the public Internet
  - Location determination methods are out of scope
- Work quite mature already. A number of RFCs associated with this work are already available.
  - RFC 4079 (architecture framework), RFC4119 (Data Format)

GEOPRIV Objectives and Requirements

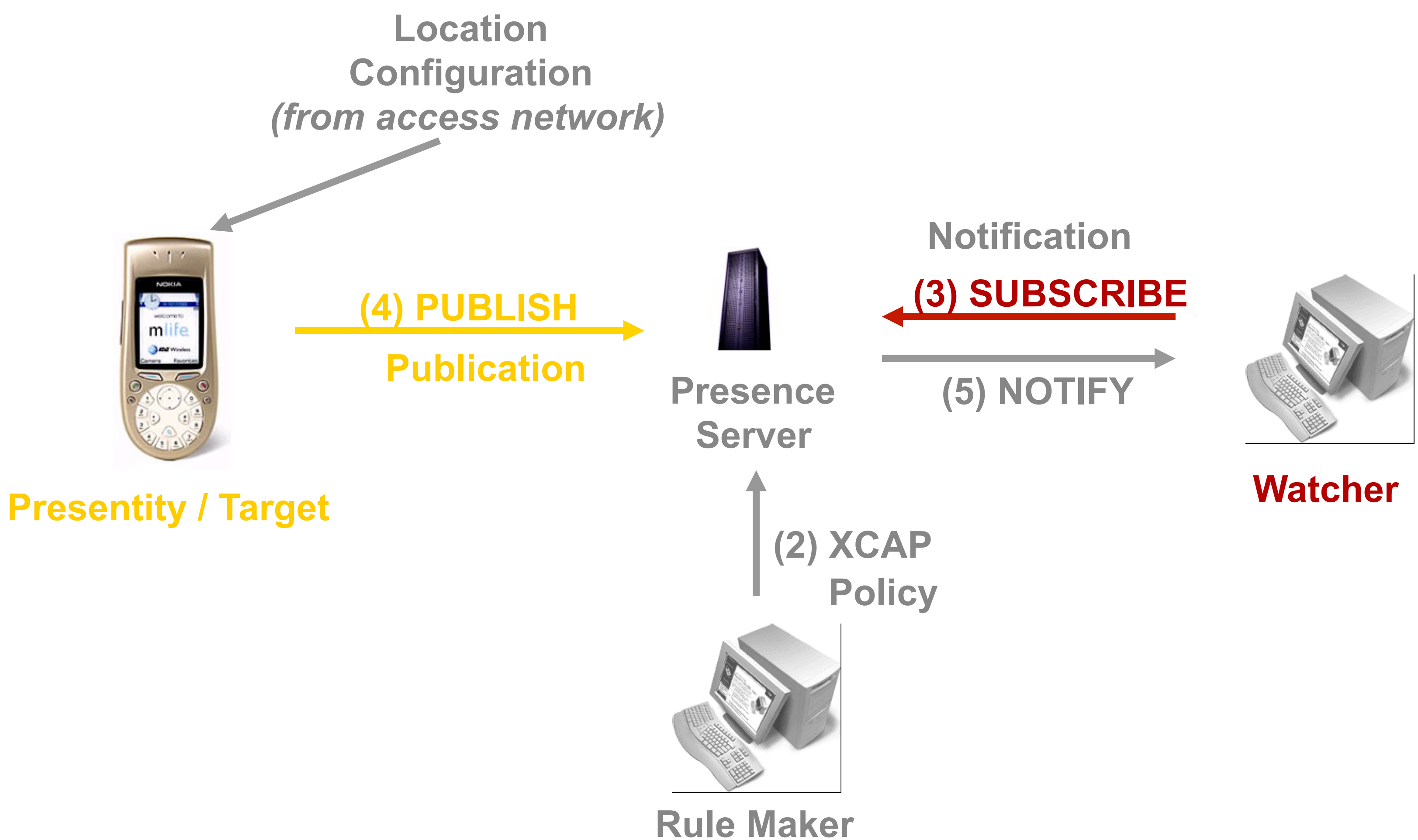
- Identify using protocols and document format for **carrying location information**
  - Allow push model and subscription model
  - Provide strong security measures to protect location information in transit
  - Insert policy directives into location information
- Develop **authorization policy language** for distribution of location information
  - Third parties enforce policies on behalf of “rule maker”
  - Motivated by a concern that many producers of geolocation information will not be controlled by end users
  - Rule Maker may be the owner of the target device, or may not

Basic GEOPRIV Architecture (RFC4079)



Shows only the network agents, not the human actors

Basic Presence Model of IETF SIMPLE Instantiating the GEOPRIV model



Geolocation and Presence

- Geopriv
  - Real-time information, changing frequently
  - Requires subscription model
  - Use servers to enforce policy
  - Need to be able to share information selectively
  - Strong authentication & confidentiality model
  - Extensibility (XML) required

- Presence
  - Ditto
  - Ditto
  - Ditto
  - Ditto
  - Ditto
  - Ditto

The Protocol: Schemas for Location Information

- The IETF does not want to define location information formats
  - Experts on these matters are largely elsewhere
- Instead, the IETF is focusing on architectures and tools for the secure distribution of location information documents
- Defining an envelope to carry any XML-based location information format
  - Popular choice is Geographic Markup Language (GML) (from OCG)
- No standardized format for civic location was available
  - Developed in Geopriv working group

PIDF-LO: RFC 4119

- Presence Information Data Format (PIDF) is an XML-based format for presence (RFC 3863)
- Extends PIDF to accommodate two new elements:
  - Location-Info
    - Encapsulates location information
    - GML 3.0 <feature.xsd> schema (mandatory-to-implement)
    - Supports civic location format (optional-to-implement)
  - Usage-rules
    - Used to indicate privacy preferences