Chapter 3: IP Multimedia Subsystems and Application-Level Signaling

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Outline

3.1 Signaling in IP Networks

3.2 3GPP IP Multimedia Subsystem (IMS)

3.3 3GPP2 IP Multimedia Subsystem (IMS)

3.1 Signaling in IP Networks

- 3.1.1 Session Initiation Protocol (SIP)
- 3.1.2 Session Description Protocol (SDP)

3.1.1 Session Initiation Protocol (SIP)

- SIP is an application-layer protocol that can establish, modify and terminate multimedia sessions (conferences) over the Internet.
- SIP messages could contain session descriptions such that participants can negotiate with media types and other parameters of the session.
- SIP provides its own mechanisms for reliability and can run on top of several different transport protocols such as TCP, UDP and SCTP (Stream Control Transmission Protocol).

SIP Capabilities

- Determine destination user's current location
- Determine whether a user is willing to participate in a session
- Determine the capabilities of a user's terminal.
- Set up a session
- Manage a session. This includes modifying the parameters of a session, invoking service functions to provide services to a session, and terminating of a session.

SIP Components

- SIP user agent (UA)
  - user agent client (UAC)
  - user agent server (UAS)
- SIP redirect server: UAS
- SIP proxy server: UAC and UAS
- SIP registrar: UAS
  - Location service

SIP

- 3.1.1.1 Naming and Addressing
- 3.1.1.2 Messages
- 3.1.1.3 Location Registration
- 3.1.1.4 Session Establishment and Termination

3.1.1.1 Naming and Addressing

- SIP Uniform Resource Identifier (URI)
  - sip:tao@research.telcordia.com
  - sips:tao@research.telcordia.com
  - sip:user:password@host:port;uri-parameters?headers
  - sip:+886-3-574-2961:1234@cs.nthu.edu.tw;user=phone
  - sip:jcchen@cs.nthu.edu.tw?subject=Wiley%20Book&priority=urgent
  - sip:wire.cs.nthu.edu.tw

URI Parameters

- parameter-name=parameter-value
- transport: UDP, TCP, SCTP, TLS, etc.
  - transport=udp is equivalent to Transport=UDP
- maddr: indicate a proxy that must be traversed to the destination
  - maddr=140.114.79.60
- ttl: used only when the maddr is a multicast address and the transport protocol is UDP
- user: distinguish a real telephone number from a user name that resembles a telephone number
- method: specifies the method of the SIP URI request
- lr: used when a specific SIP routing mechanism is implemented (will not discuss further)

3.1.1.2 Messages

- **INVITE:** Used by a user to invite another user to establish a SIP session
- **ACK:** Used to confirm final response
- **BYE:** Used to terminate a session
- **CANCEL:** Used to cancel a SIP request
- **OPTIONS:** Used to query servers about their capabilities
- **REGISTER:** Used by a user to register information with a server
- **INFO:** Used to carry session related control information
- **SUBSCRIBE:** Used to request current state and state updates from a remote node
- **NOTIFY:** Used to notify a SIP node that an event which has been requested by an earlier SUBSCRIBE method has occurred
- **PRACK:** Used to provide a reliable Provisional Response ACKnowledgement
- **UPDATE:** Used to update parameters of a session
- **MESSAGE:** Used to transfer Instant Messages (IM)
- **REFER:** Used to direct a recipient to other resource by using the contact information provided in the REFER request

Message Format

- A start-line
  - Request-Line
  - Status-Line
- One or more header fields
- An empty line indicating the end of the header fields
- An optional message body

Table 3.1 Structure of a SIP message

Start-line	INVITE sip:tao@research.telcordia.com SIP/2.0
Header Field(s)	Via: SIP/2.0/UDP fly.cs.nthu.edu.tw:5060;branch=z9hG4bK776asdhds Max-Forwards: 70 To: Tao <sip:tao@research.telcordia.com> From: Jyh-Cheng <sip:jcchen@cs.nthu.edu.tw>;tag=1928301774 Call-ID: a84b4c76e66710@fly.cs.nthu.edu.tw CSeq: 123456 INVITE Contact: <sip:jcchen@fly.cs.nthu.edu.tw> Content-Type: application/sdp Content-Length: 132
Empty Line	
Message Body (Optional)	v=0 t=2873397496 2873404696 m=audio 49170 RTP/AVP 0 ……

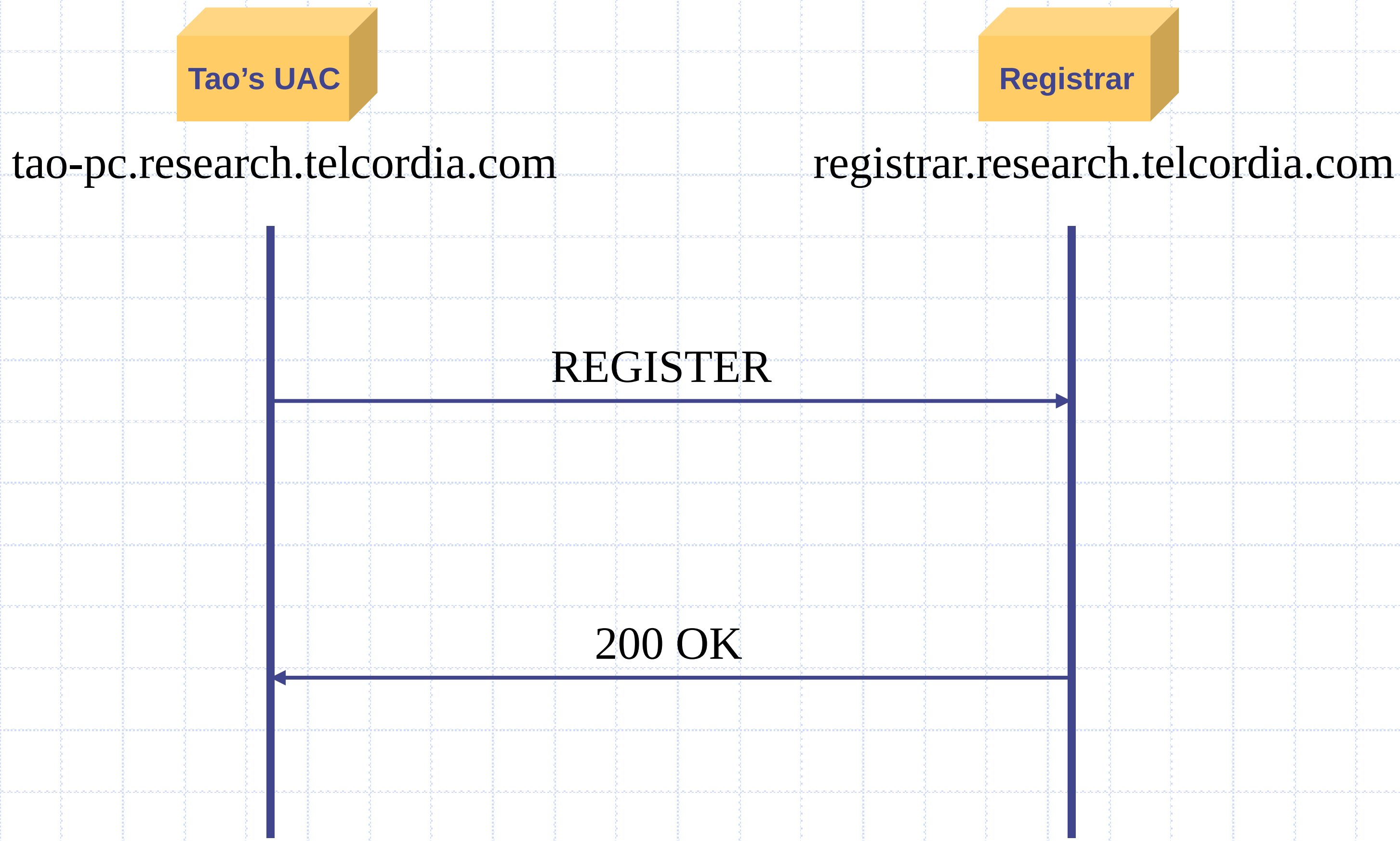
Status-Line

- Status-Code
  - 1xx: Provisional – indicate a request is received and is being processed.
  - 2xx: Success – indicate the method invoked by a request is successfully accepted.
    - E.g. SIP/2.0 200 OK
  - 3xx: Redirection – further action needs to be taken by the sender of the corresponding sender in order to complete the request.
  - 4xx: Client error – the request contains syntax error or cannot fulfilled at this server.
  - 5xx: Server error – the server failed to fulfill an apparently valid request.
  - 6xx: Global failure – the request cannot be fulfilled at any server.

3.1.1.3 Location Registration

- Address of the registrar
  - Preconfigured
  - address-of-record
    - *sip:tao@research.telcordia.com* will send REGISTER to *sip:research.telcordia.com*
  - Multicast address
    - In IPv4, 224.0.1.75 has been allocated to sip.mcast.net

Fig. 3.1 SIP registration



Example of REGISTER

REGISTER sip:registrar.research.telcordia.com SIP/2.0
Via: SIP/2.0/UDP tao-pc.research.telcordia.com:5060;
branch=z9hG4bKnashds7
Max-Forwards: 70
To: Tao <sip:tao@research.telcordia.com>
From: Tao <sip:tao@research.telcordia.com>
Call-ID: 843817638423076@989sddhas09
CSeq: 2660 REGISTER
Contact: <sip:tao@128.96.60.187>
Expires: 3600
Content-Length: 0

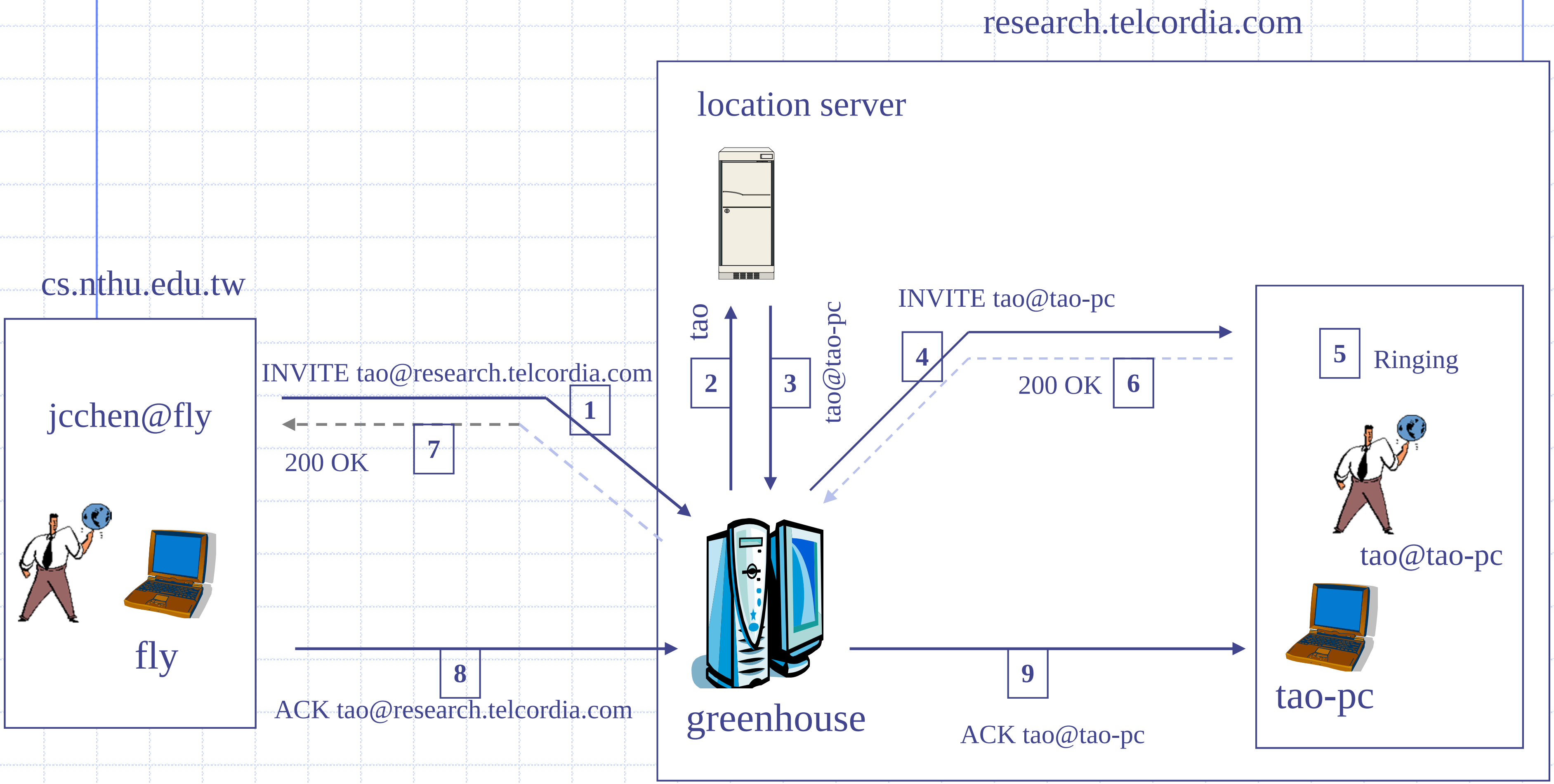
Example of OK

SIP/2.0 200 OK
Via: SIP/2.0/UDP tao-pc.research.telcordia.com:5060;
  branch=z9hG4bKnashds7;received=128.96.60.187
To: Tao <sip:tao@research.telcordia.com>
From: Tao <sip:tao@research.telcordia.com>
Call-ID: 843817638423076@989sddhas09
CSeq: 2660 REGISTER
Contact: <sip:tao@128.96.60.187>
Expires: 3600
Content-Length: 0

3.1.1.4 Session Establishment and Termination

- Peer-to-peer mode
  - a caller establishes a call to a callee directly without going through any SIP server
- Server mode
  - Proxy server
    - forward the received SIP request toward its final destination on behave of the originator
    - may rewrite specific parts of the message before forwarding it
  - Redirect server
    - respond to a request with the callee's contact information to indicate where the caller should contact next

Fig. 3.2 SIP in proxy mode



INVITE

INVITE sip:tao@research.telcordia.com SIP/2.0
Via: SIP/2.0/UDP
    fly.cs.nthu.edu.tw:5060;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Tao <sip:tao@research.telcordia.com>
From: Jyh-Cheng <sip:jcchen@cs.nthu.edu.tw>;tag=1928301774
Call-ID: a84b4c76e66710@fly.cs.nthu.edu.tw
CSeq: 123456 INVITE
Contact: <sip:jcchen@fly.cs.nthu.edu.tw>
Content-Type: application/sdp
Content-Length: 132

200 OK

SIP/2.0 200 OK
Via: SIP/2.0/UDP greenhouse.research.telcordia.com:5060;
    branch=z9hG4bKnashds8;received=207.3.230.150
Via: SIP/2.0/UDP fly.cs.nthu.edu.tw:5060;
    branch=z9hG4bK776asdhds;received=140.114.79.59
To: Tao <sip:tao@research.telcordia.com>;tag=a6c85cf
From: Jyh-Cheng <sip:jcchen@cs.nthu.edu.tw>;tag=1928301774
Call-ID: a84b4c76e66710@fly.cs.nthu.edu.tw
CSeq: 123456 INVITE
Contact: <sip:tao@tao-pc.research.telcordia.com>
Content-Type: application/sdp
Content-Length: 121

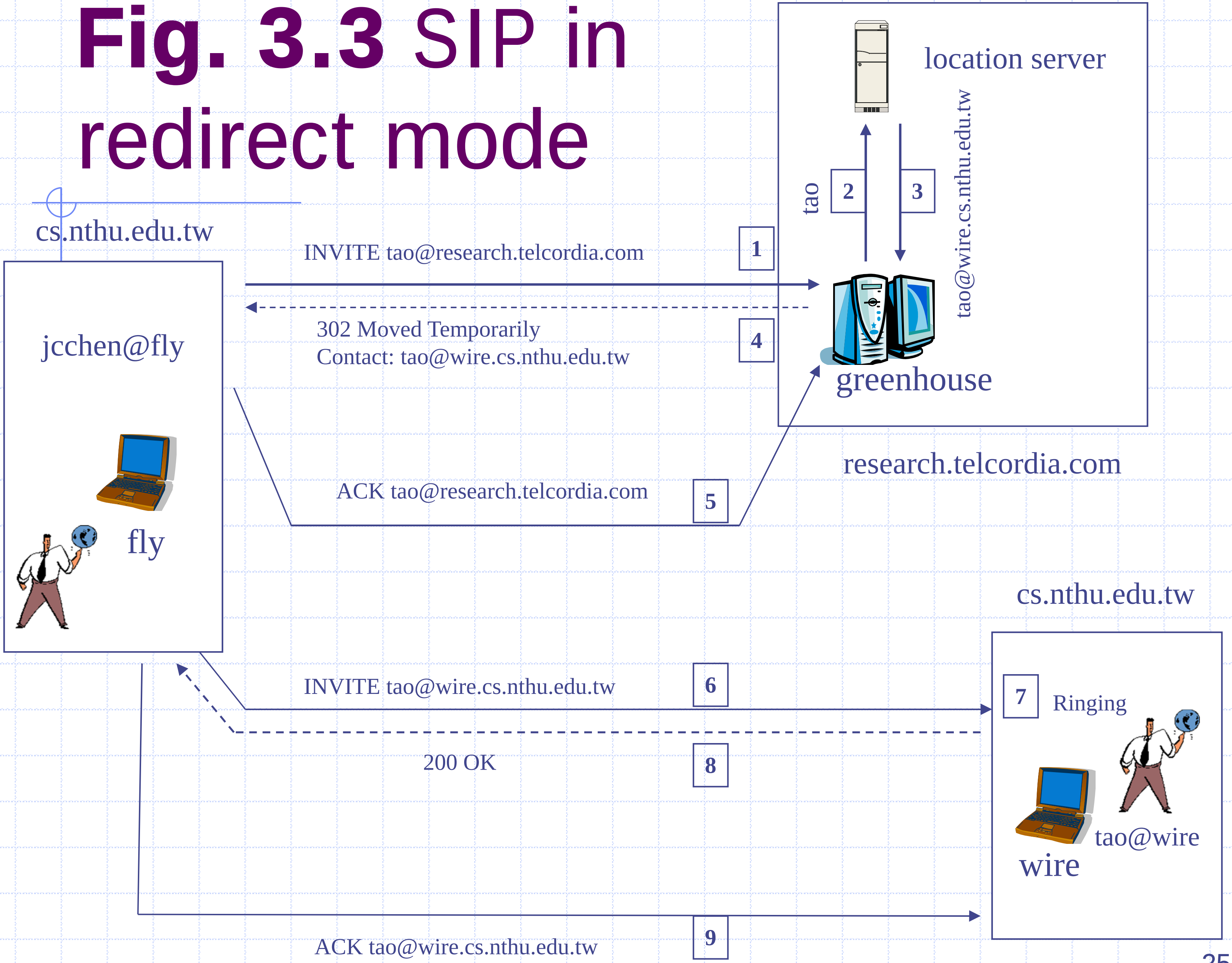
ACK

ACK sip:tao@research.telcordia.com SIP/2.0
Via: SIP/2.0/UDP fly.cs.nthu.edu.tw:5060;
  branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Tao <sip:tao@research.telcordia.com>;
  tag=a6c85cf
From: Jyh-Cheng <sip:jcchen@cs.nthu.edu.tw>;
  tag=1928301774
Call-ID: a84b4c76e66710@fly.cs.nthu.edu.tw
CSeq: 123456 ACK
Content-Length: 0

BYE

```
BYE sip:jcchen@cs.nthu.edu.tw SIP/2.0
Via: SIP/2.0/UDP tao-pc.research.telcordia.com;
   branch=z9hG4bKnashds10
Max-Forwards: 70
From: Tao <sip:tao@research.telcordia.com>;
   tag=a6c85cf
To: Jyh-Cheng <sip:jcchen@cs.nthu.edu.tw>;
   tag=1928301774
Call-ID: a84b4c76e66710@fly.cs.nthu.edu.tw
CSeq: 231 BYE
Content-Length: 0
```

Fig. 3.3 SIP in redirect mode



3.1.2 Session Description Protocol (SDP)

- Designed to describe multimedia sessions
  - convey information of media streams so prospective participants of multimedia sessions could learn the relevant setup information
- Does not incorporate any transport protocol
  - a common usage of SDP is to embed SDP in the payload of other protocols

Format

<type>=<value>

- Name and purpose of the session
- Activation time of the session
- Media comprising the session
- Information, such as address, port number, and format, to receive the media

Example

```
v=0
o=jcchen 2890844526 2890842807 IN IP4
   140.114.79.59
s=Wiley Book
i=Discussion on book writing
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
m=application 32416 udp wb
```

Offer/Answer Model

- For unicast
- To find common codecs both participants can support
- Either one of the participants may generate a new offer message to update the session
- Mandatory for SIP

Offer (in INVITE)

v=0
o=Jyh-Cheng 2890844526 2890844526 IN IP4 fly.cs.nthu.edu.tw
s=
c=IN IP4 fly.cs.nthu.edu.tw
t=0 0
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
m=application 32416 udp wb

Answer (in 200 OK)

v=0

o=Tao 2890844730 2890844730 IN IP4 tao-pc.research.telcordia.com

s=

c=IN IP4 tao-pc.research.telcordia.com

t=0 0

m=audio 49920 RTP/AVP 0

m=video **0** RTP/AVP 31

m=application 32416 udp wb

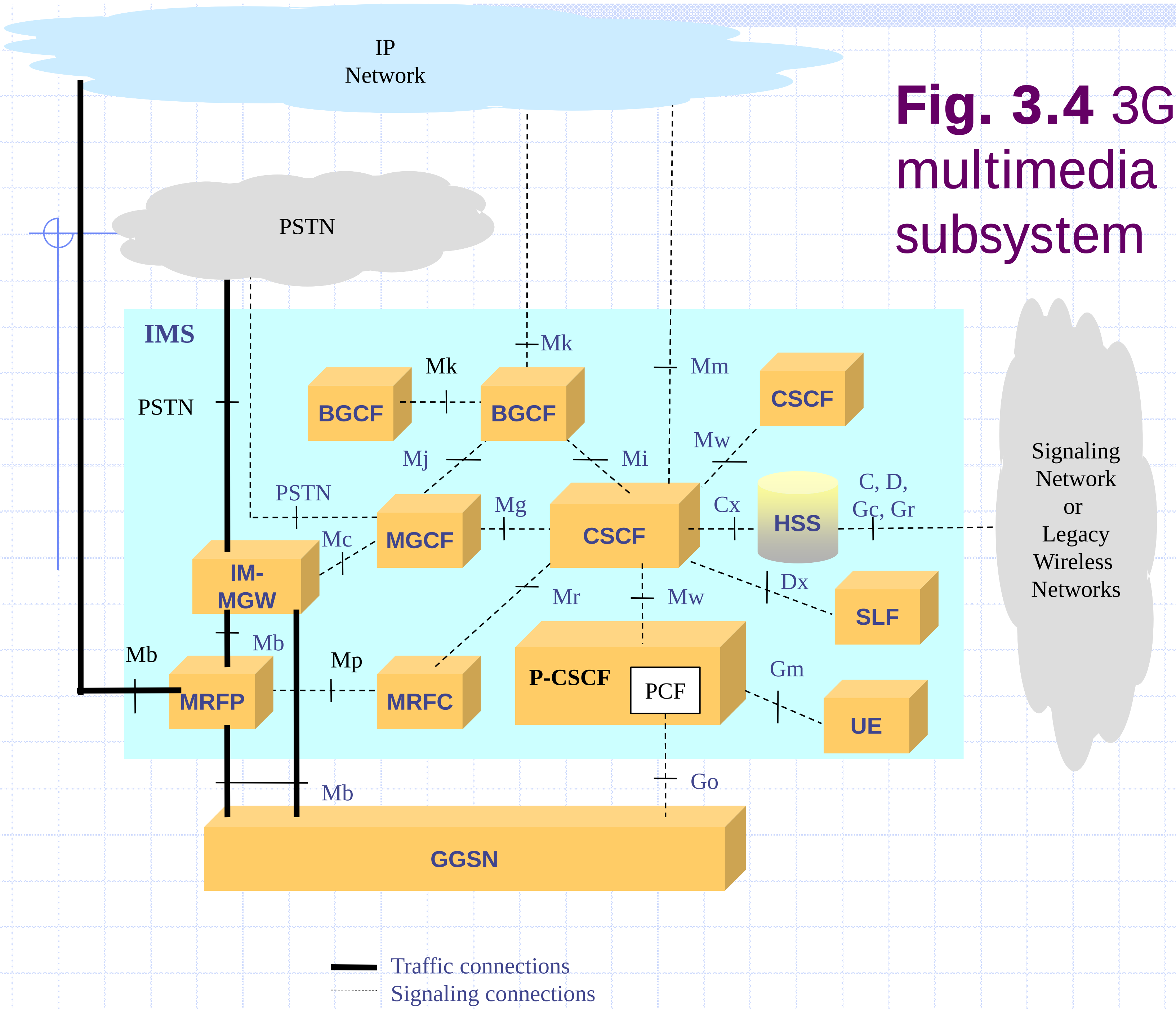
3.2 3GPP IP Multimedia Subsystem (IMS)

- 3.2.1 IMS Architecture
- 3.2.2 Mobile Station Addressing for Accessing the IMS
- 3.2.3 Reference Interfaces
- 3.2.4 Service Architecture
- 3.2.5 Registration with the IMS
- 3.2.6 Deregistration with the IMS
- 3.2.7 End-to-End Signaling Flows for Session Control

3.2.1 IMS Architecture

- Support real-time voice and multimedia IP applications
- Use SIP to support signaling and session control
- Call State Control Function (CSCF): a SIP server
  - Serving CSCF (S-CSCF)
  - Proxy CSCF (P-CSCF)
  - Interrogating CSCF (I-CSCF)

Fig. 3.4 3GPP IP multimedia subsystem



Serving CSCF (S-CSCF)

- Registration: A S-CSCF can act as a SIP Registrar to accept users' SIP registration requests and make users' registration and location information available to location servers such as the HSS.
- Session Control: A S-CSCF can perform SIP session control functions for a registered user.
- Proxy Server: A S-CSCF may act as a SIP Proxy Server that relays SIP messages between users and other CSCFs or SIP servers.
- Interactions with Application Servers: A S-CSCF acts as the interface to application servers and other IP or legacy service platforms.
- Other functions: A S-CSCF performs a range of other functions not mentioned above. For example, it provides service-related event notifications to users and generates Call Detail Records (CDRs) needed for accounting and billing.

Proxy CSCF (P-CSCF)

- A mobile's fist contact point inside a local (or visited) IMS
- Act as a SIP Proxy Server
  - accept SIP requests from the mobiles and then either serves these requests internally or forwards them to other servers
- Include a Policy Control Function (PCF) that controls the policy regarding how bearers in the GGSN should be used
- Perform a range of other functions

Interrogating CSCF (I-CSCF)

- An optional function that can be used to hide an operator network's internal structure from an external network
- Serve as a central contact point within an operator's network for all sessions destined to a subscriber of that network or a roaming user currently visiting that network
- Select an S-CSCF for a user's session based on
  - capabilities required by the user
  - capabilities and availability of the S-CSCFs
  - topological information such as the location of an S-CSCF and the location of the users' P-CSCFs
- Route SIP requests to the selected S-CSCF
- Generate CDRs

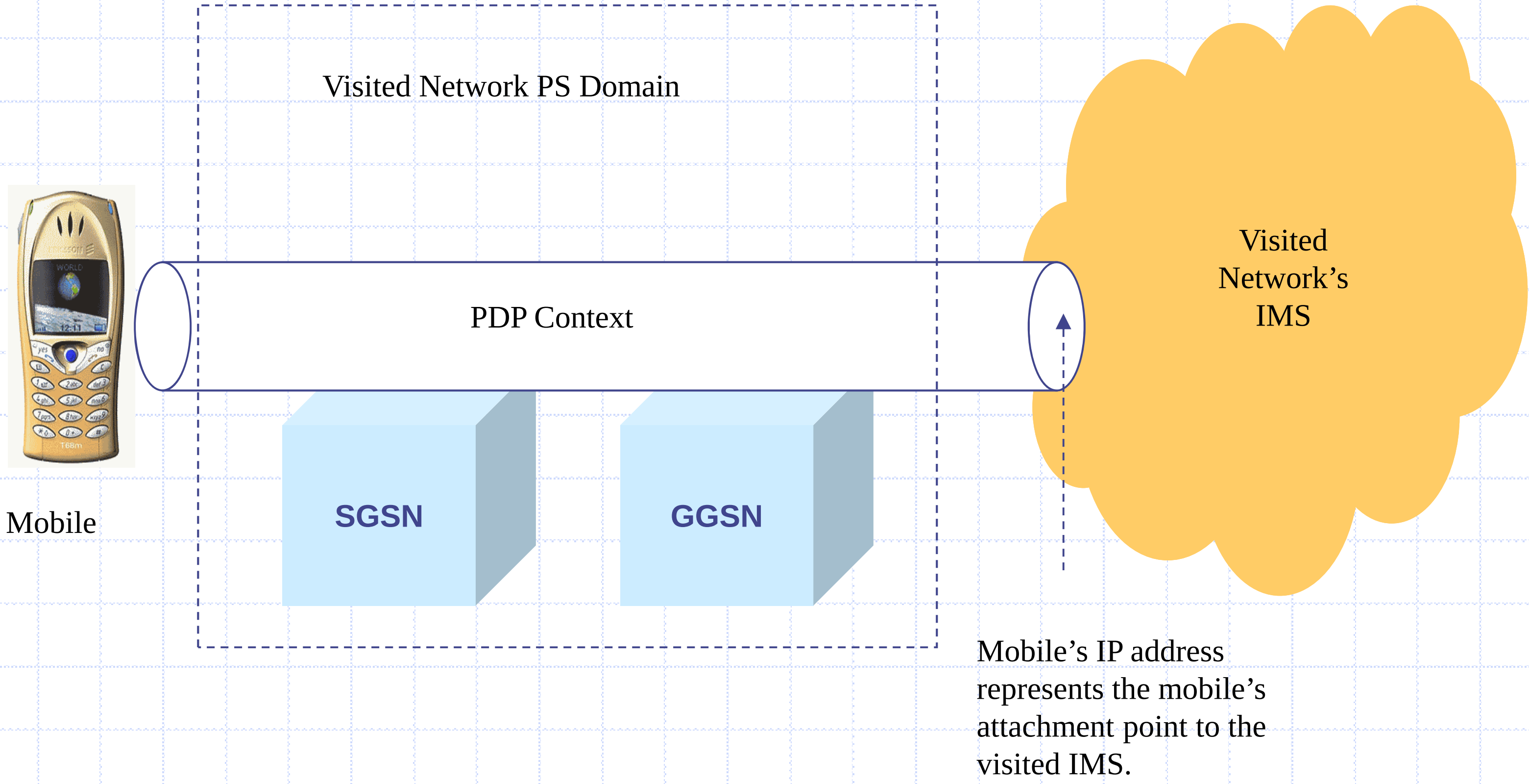
Other Components

- 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 CDRs
- 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

3.2.2 Mobile Station Addressing for Accessing the IMS

- In order for a mobile user to use the services provided by a visited IMS, the mobile needs to have an IP address (i.e., the mobile's PDP address) that is logically part of the IP addressing domain of the visited IMS.
- A PDP context will be activated for this address so that the packets addressed to this IP address can be forwarded by the 3GPP packet domain to the mobile.

Fig. 3.5 Mobile station addressing for accessing IMS services



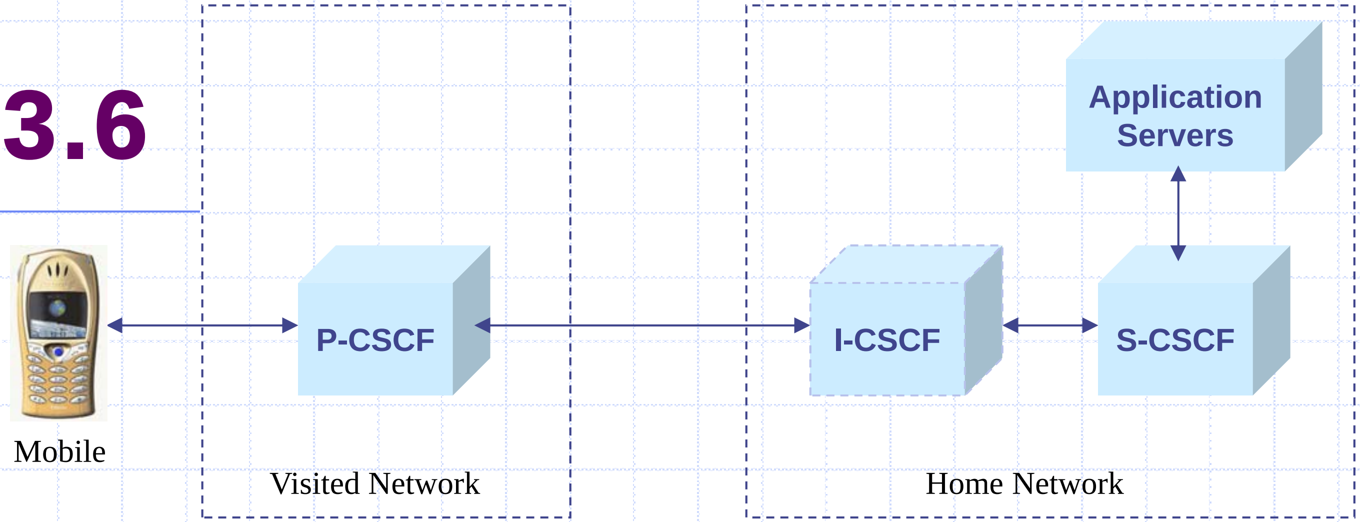
3.2.3 Reference Interfaces

- Interfaces for SIP-based signaling and service control: Mg, Mi, Mj, Mk, Mr, Mw
  - use SIP as the signaling protocol
- Interfaces for controlling media gateways: Mc, Mp
  - Use H.248 Gateway Control Protocol
- Interfaces with the Information Servers: Cx
- Interfaces with external networks: Mb, Mm, and Go
  - IP-based protocols

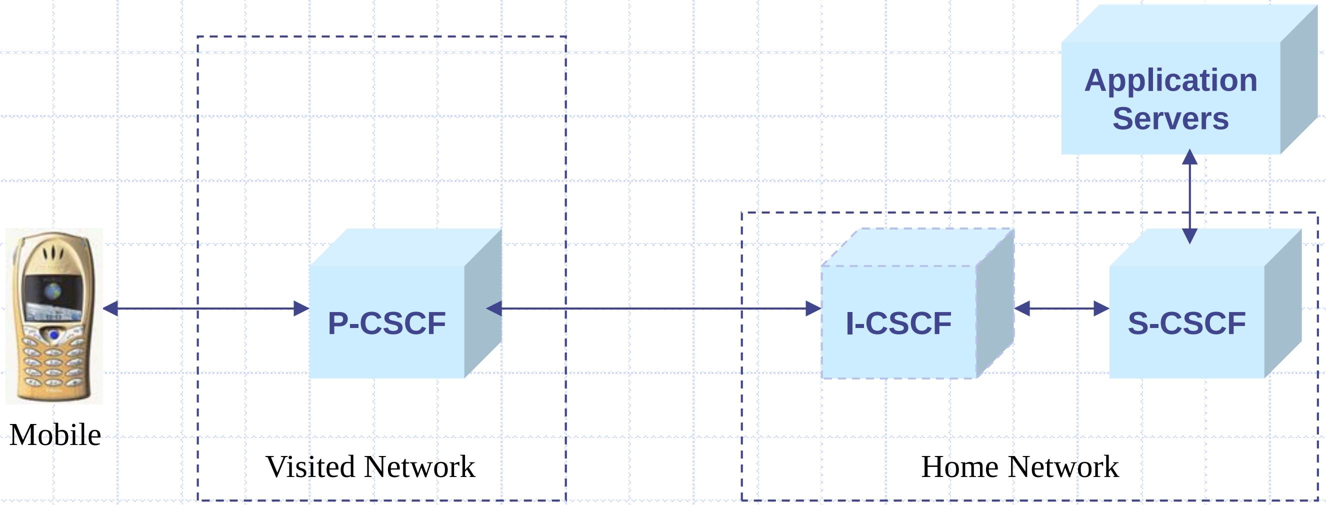
3.2.4 Service Architecture

- A mobile's home network provides service control for the mobile's *Home Subscribed Services* even when the mobile is currently in a visited network.
- A mobile's S-CSCF will always be a S-CSCF in the mobile's home network.
- A service platform provides service control for real-time services.

Fig. 3.6



(a) Service platform in mobile's home network

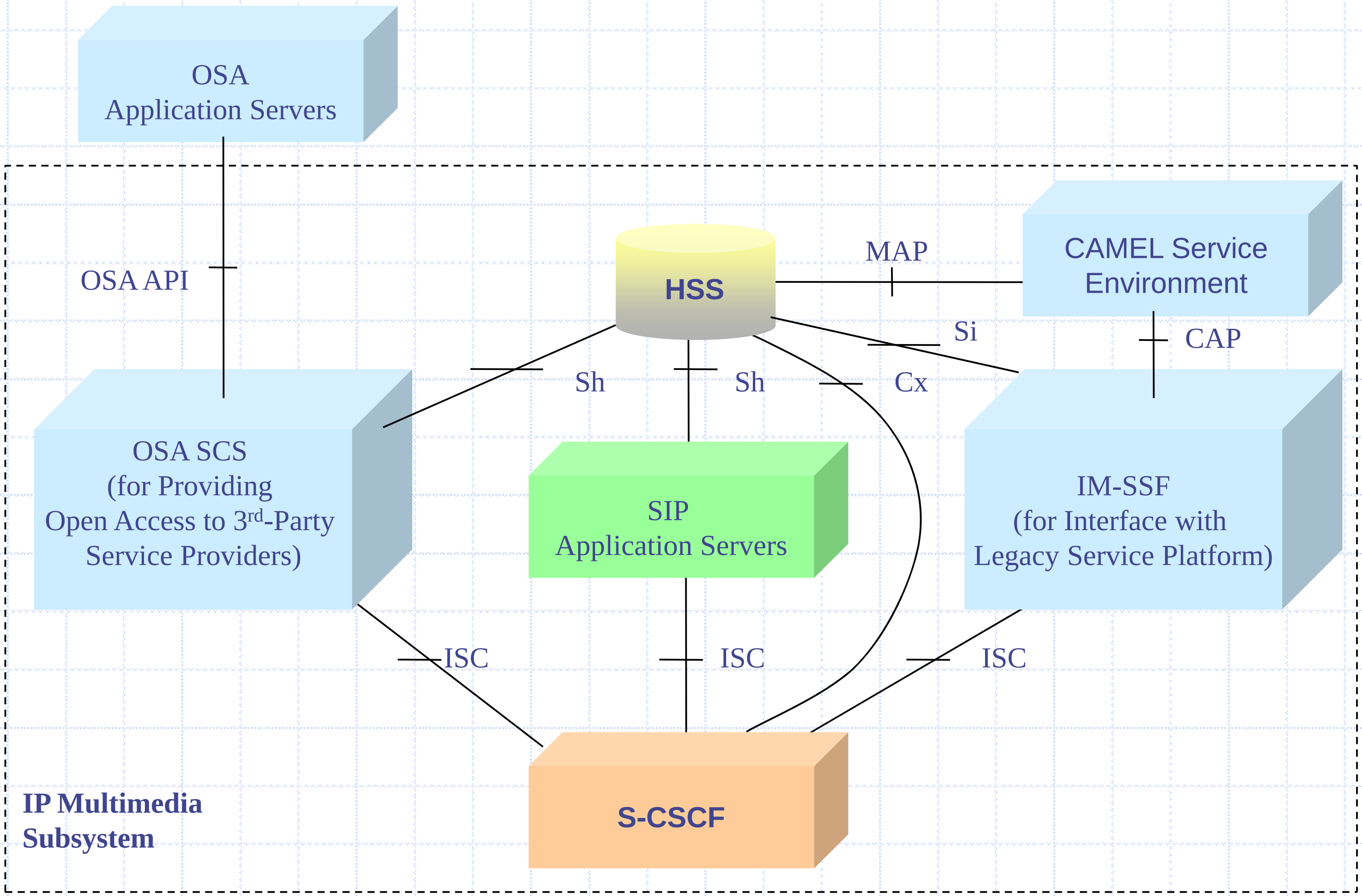


(b) External service platform

Service Platforms

- Three standardized platforms
  - SIP application server
  - Open Service Access (OSA) Service Capability Server (SCS)
    - Gateway to OSA application server
  - IP Multimedia Service Switching Function (IM-SSF)
    - Gateway to CAMEL Service Environment (CSE)
- Same interface: IMS Service Control (ISC)
  - use SIP

Fig. 3.7 Interactions between S-CSCF and service platforms



3.2.5 Registration with the IMS

- Local P-CSCF Discovery: discover the IP address of a local P-CSCF in the visited IMS
  - obtain from the visited GGSN as part of the PDP Context Activation process
  - uses DHCP to discover *after* the PDP context is activated
- Registration with IMS: perform SIP registration with the visited IMS and the mobile's home IMS

Fig. 3.8 3GPP Local P-CSCF discovery

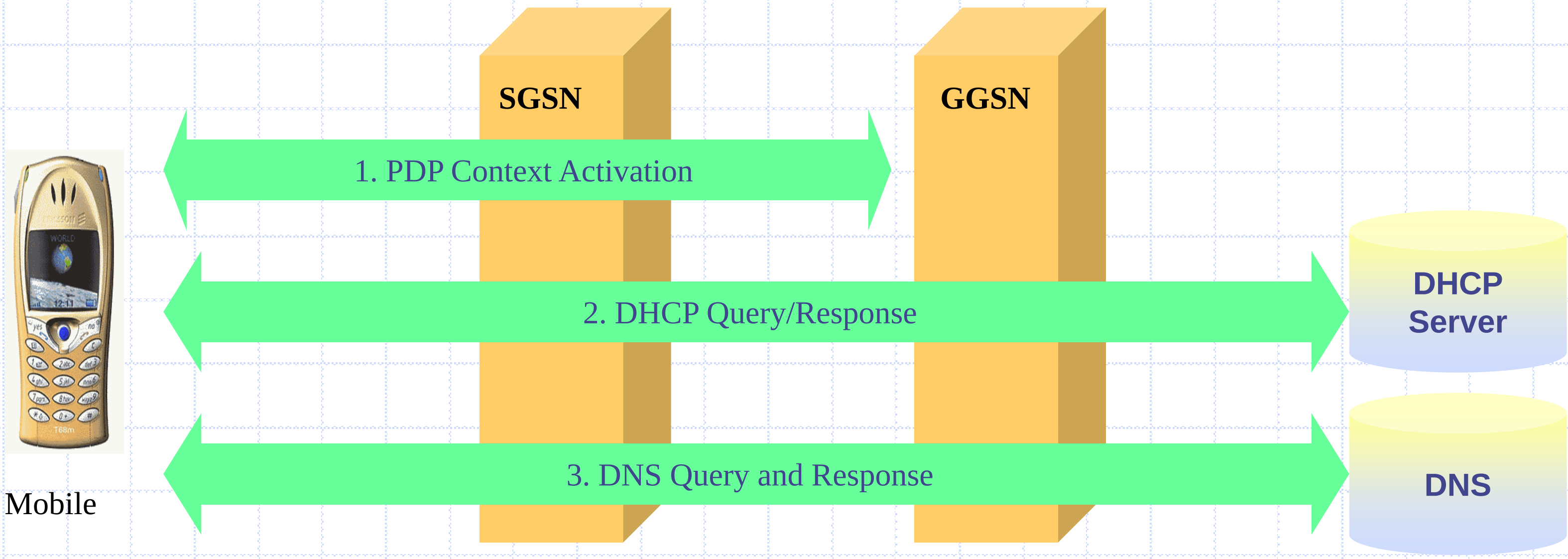
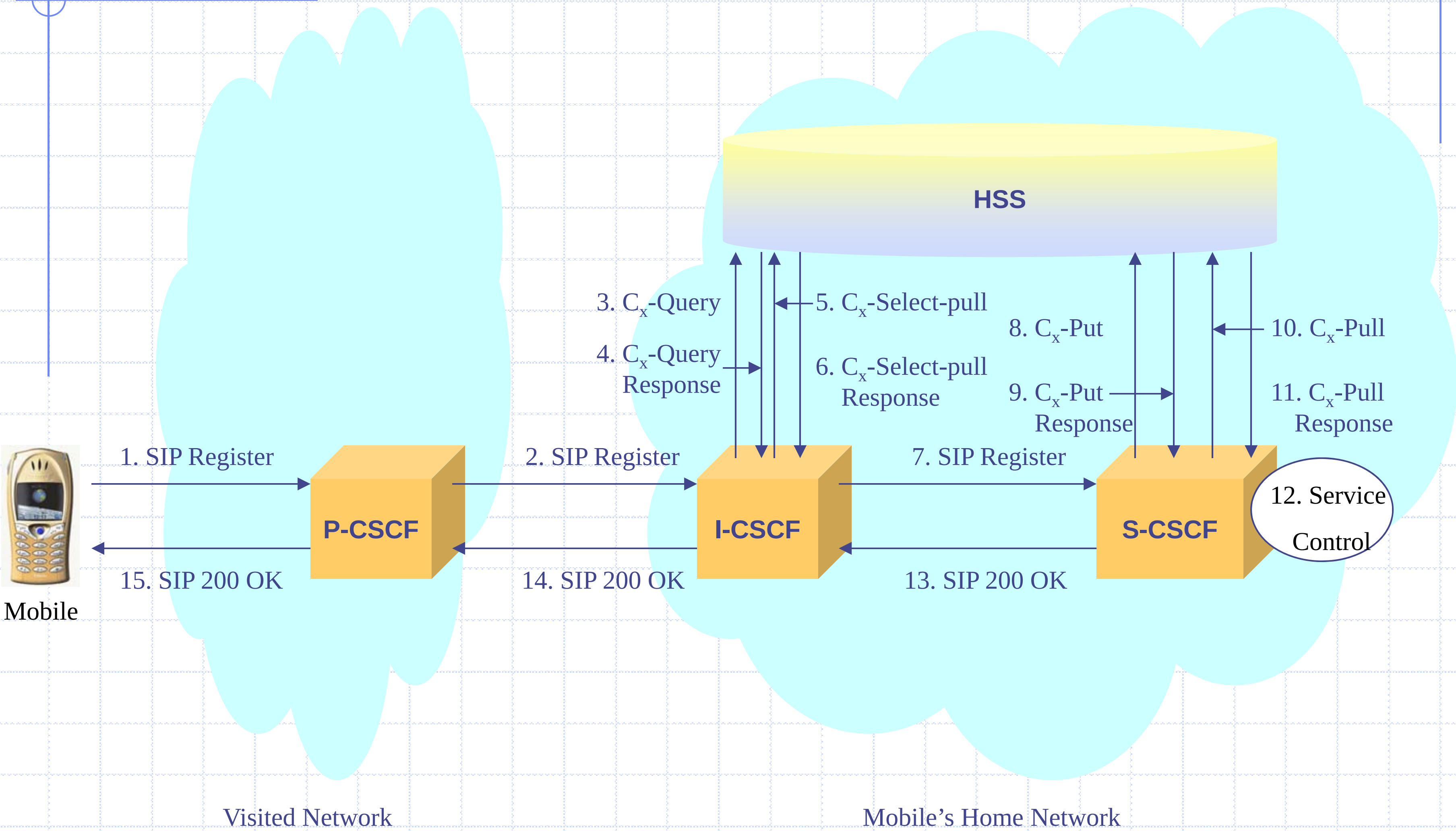


Fig. 3.9 3GPP IP Multimedia Subsystem registration procedure



3.2.6 Deregistration with the IMS

- Mobile initiated
- Network initiated
  - initiated by registration timeout
  - initiated by a network administrative function such as HSS or S-CSCF

Fig. 3.10 Mobile-initiated deregistration

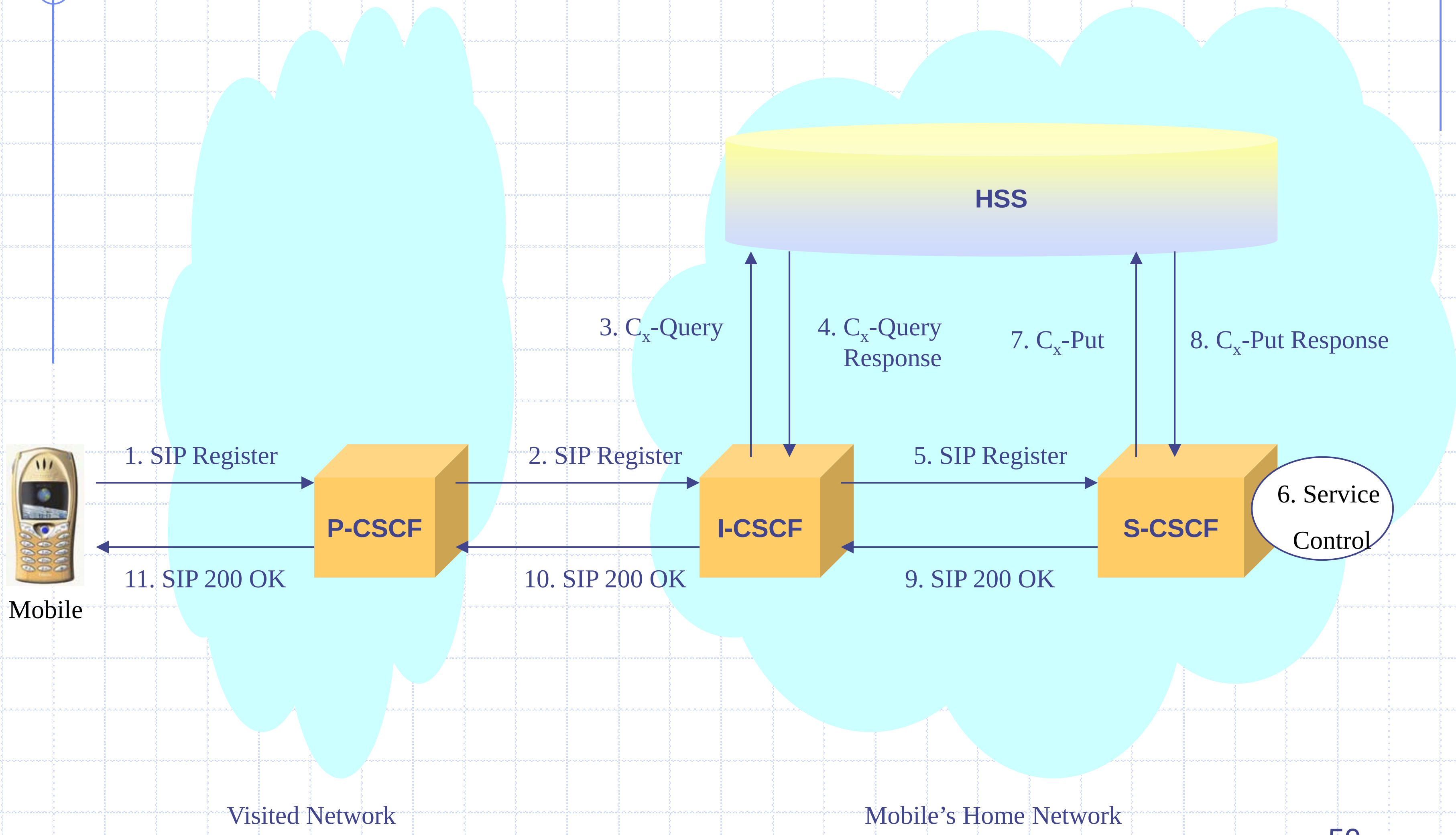


Fig. 3.11 Network-initiated deregistration by registration timeout

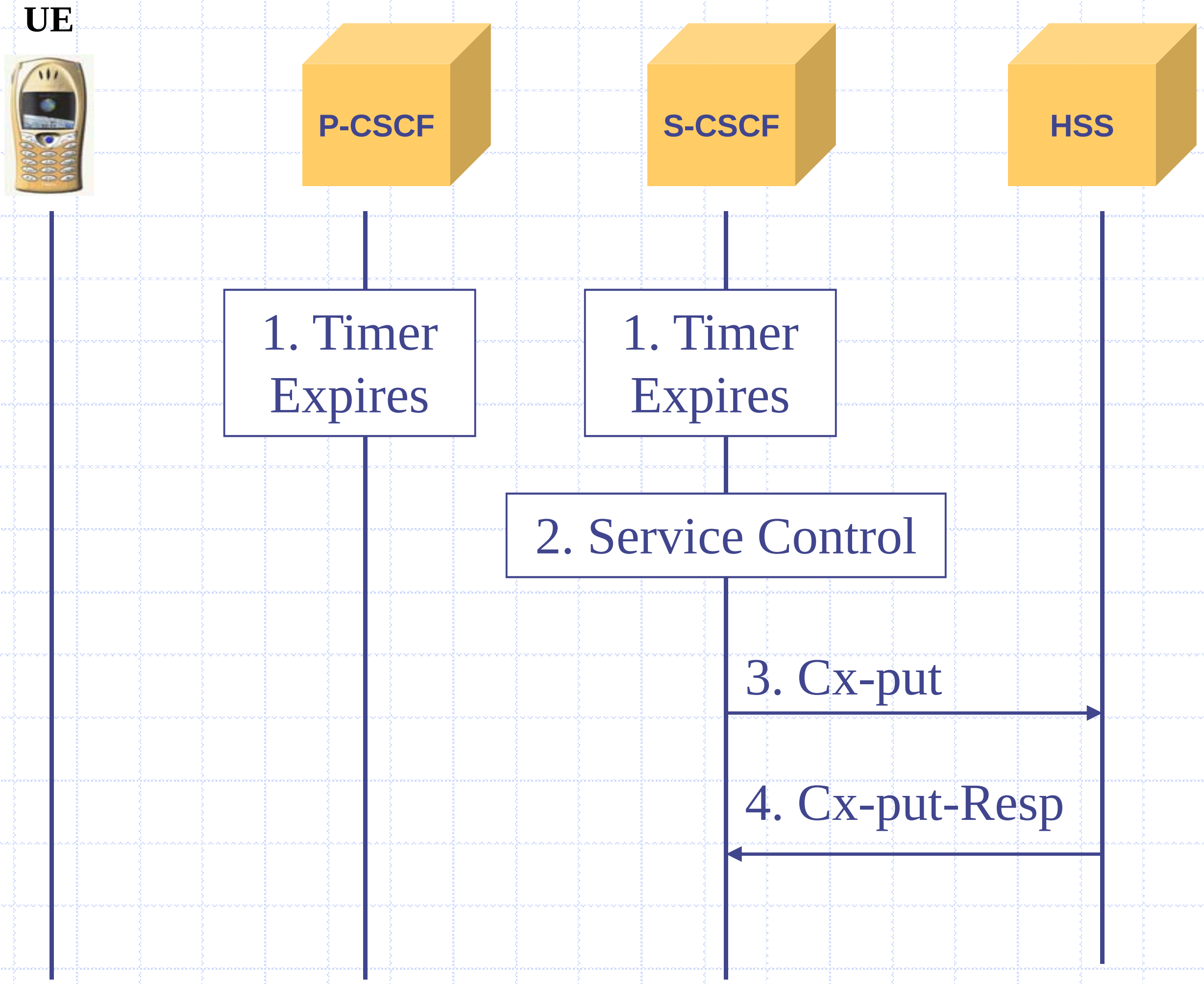


Fig. 3.12 Network-initiated deregistration by HSS

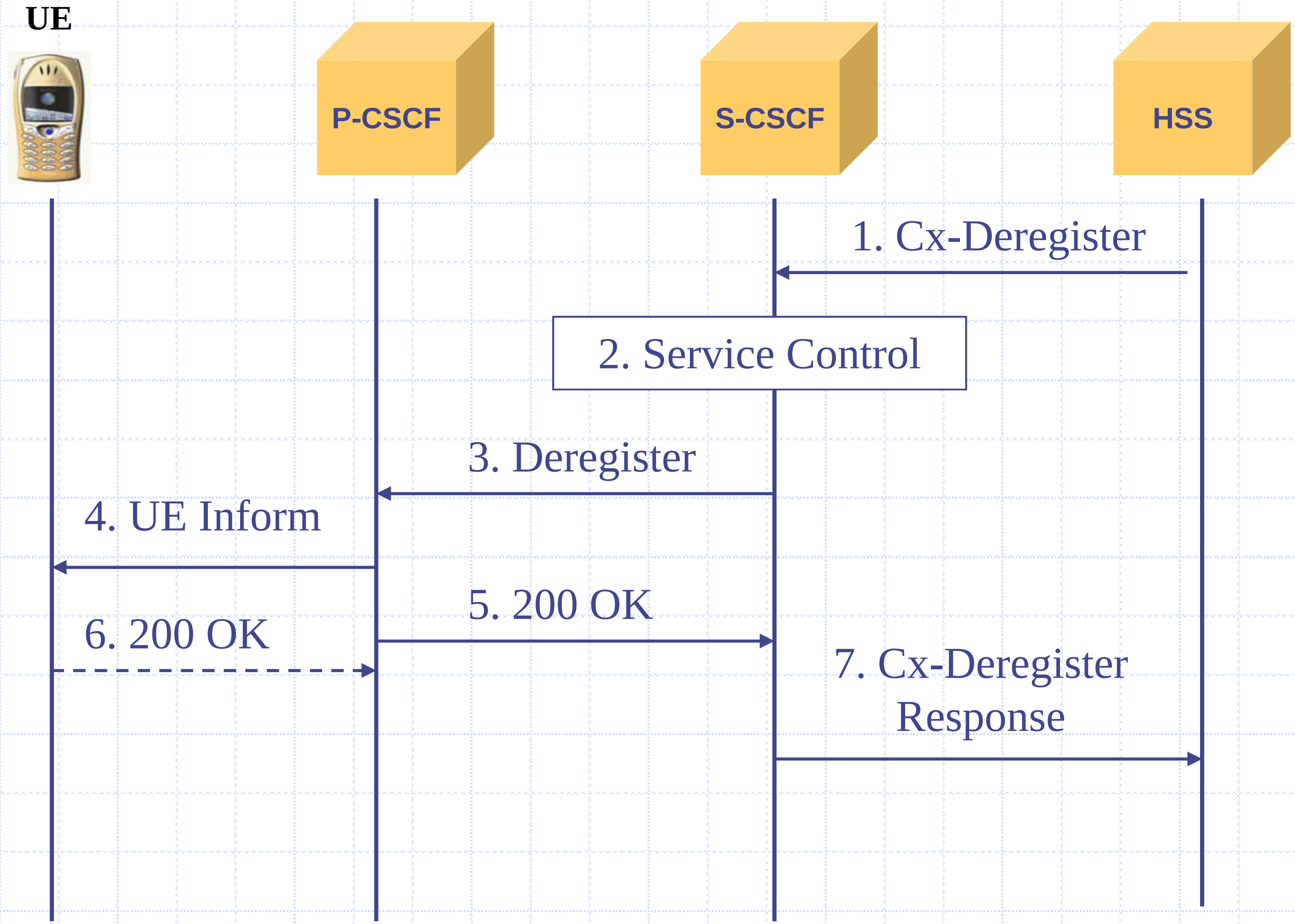
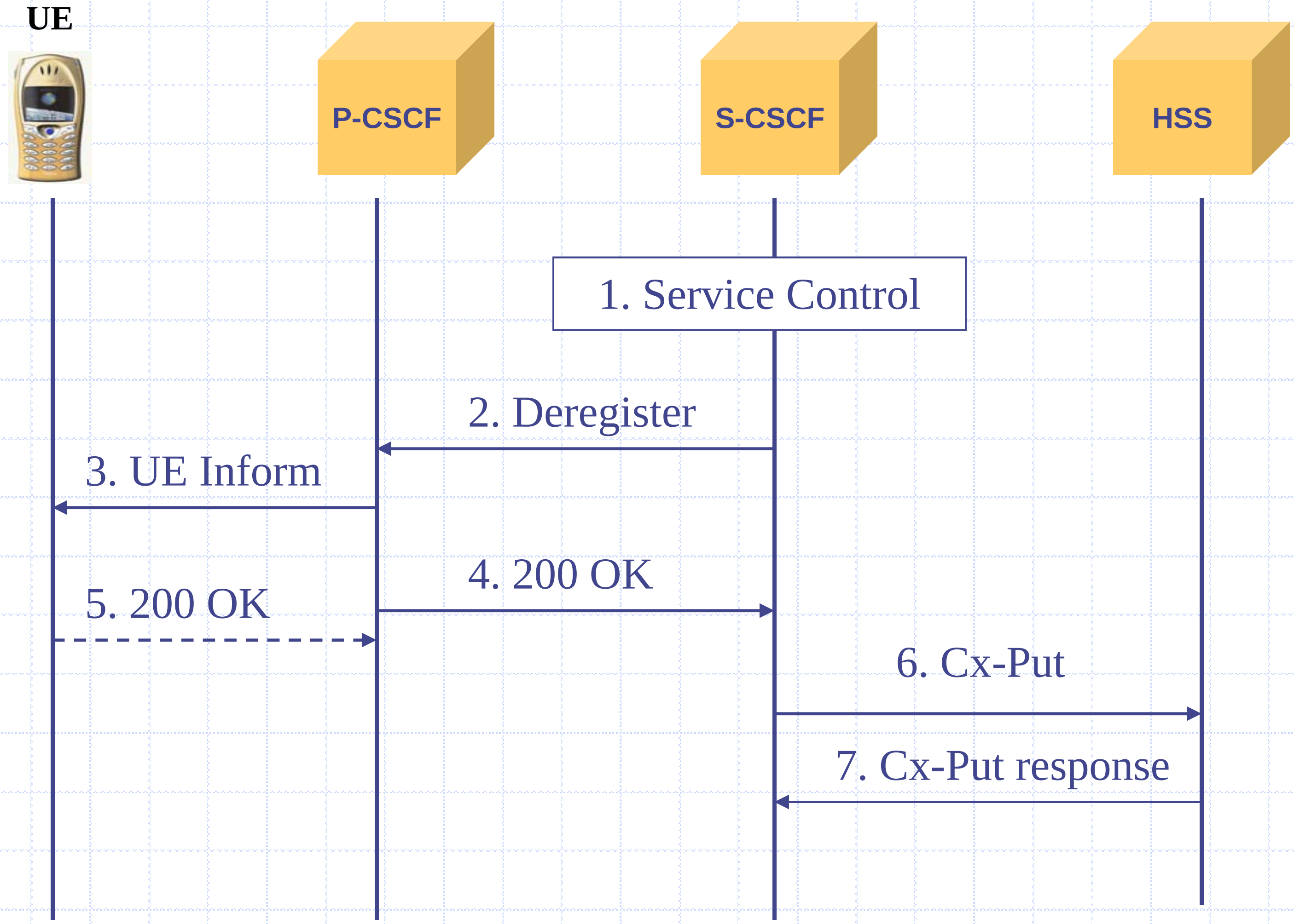


Fig. 3.13 Network-initiated deregistration by S-CSCF



3.2.7 End-to-End Signaling Flows for Session Control

- Mobile origination flow
- Mobile termination flow
- S-CSCF to S-CSCF signaling flow

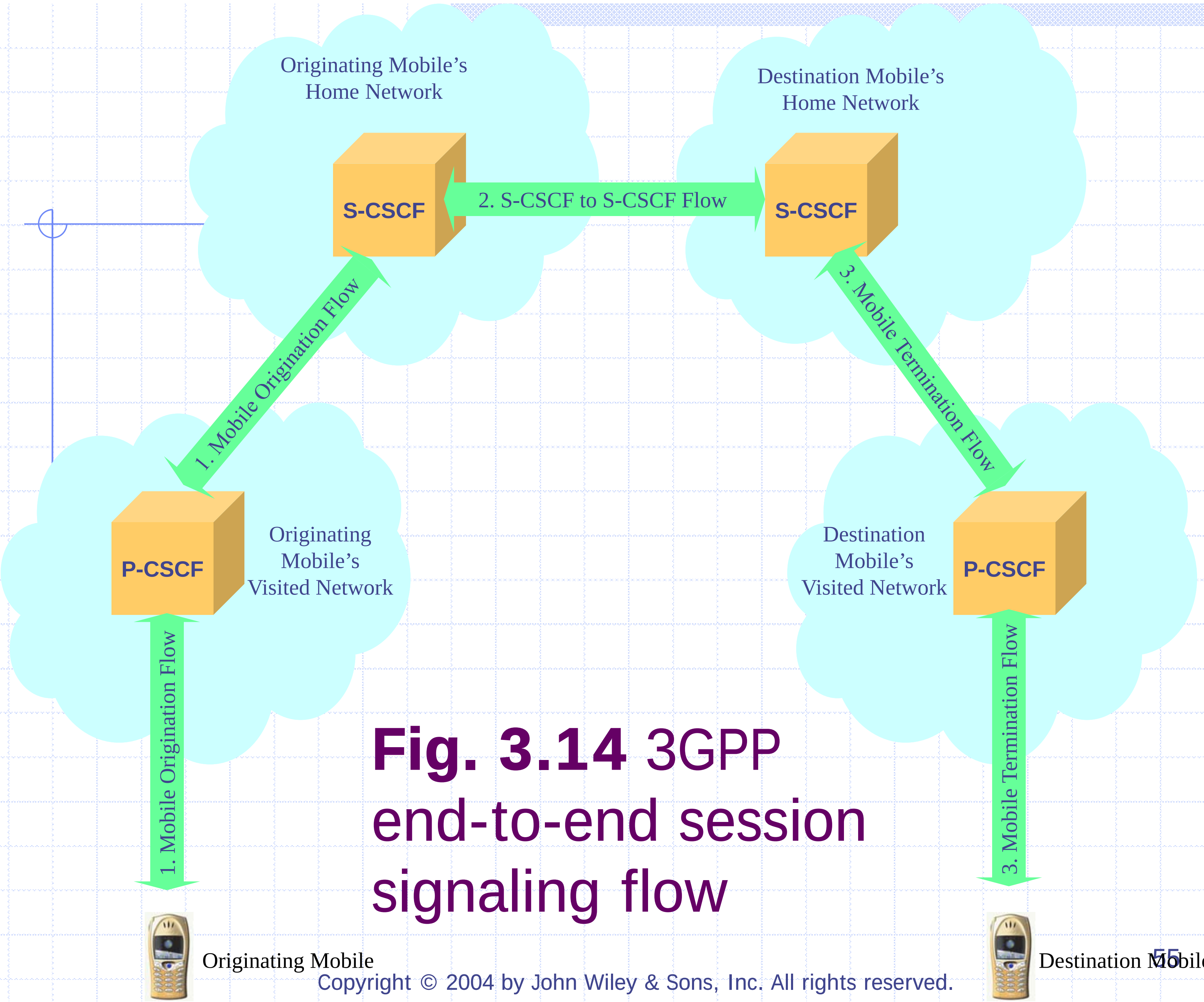


Fig. 3.14 3GPP end-to-end session signaling flow

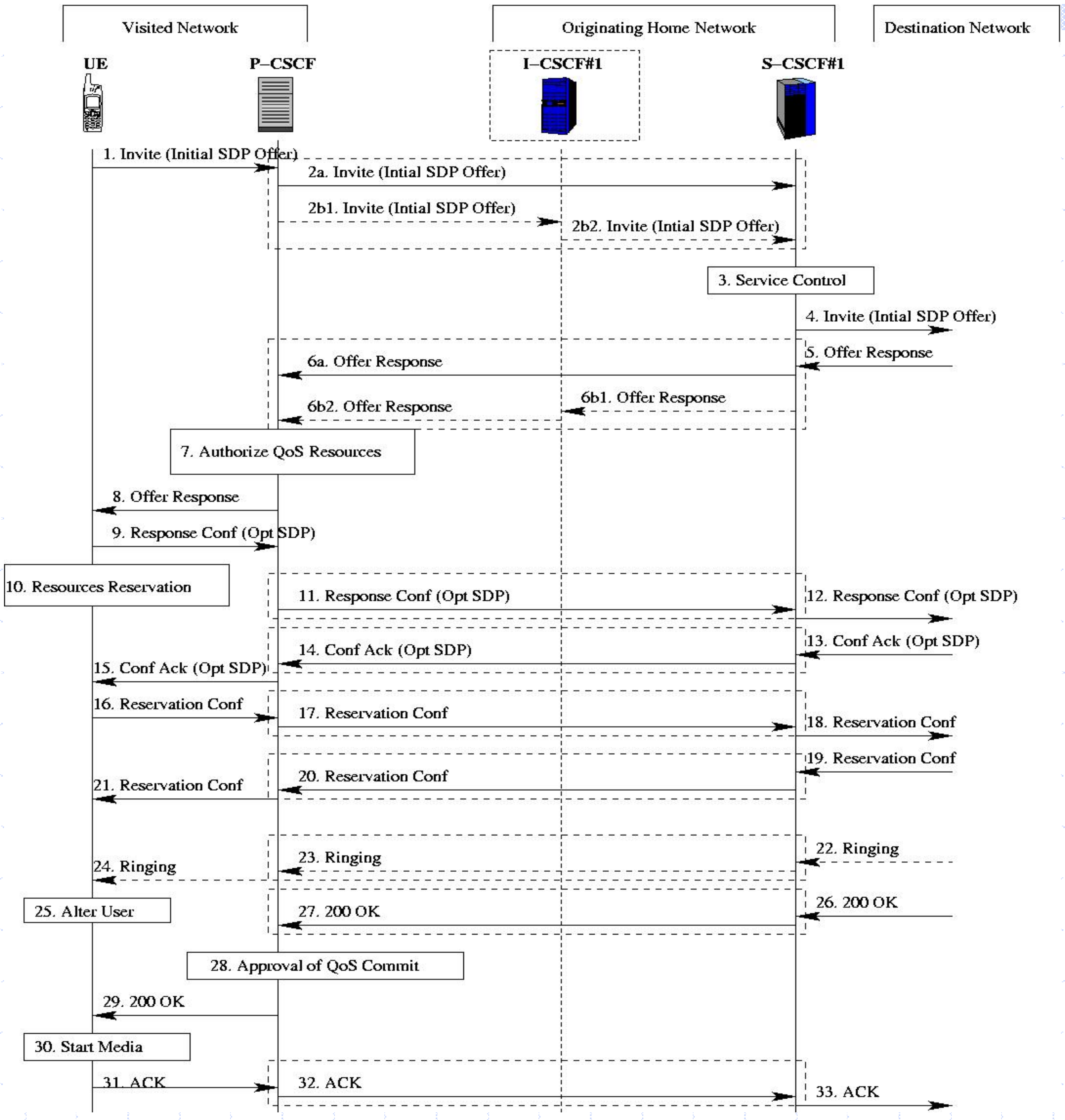


Fig. 3.15 Mobile origination flow

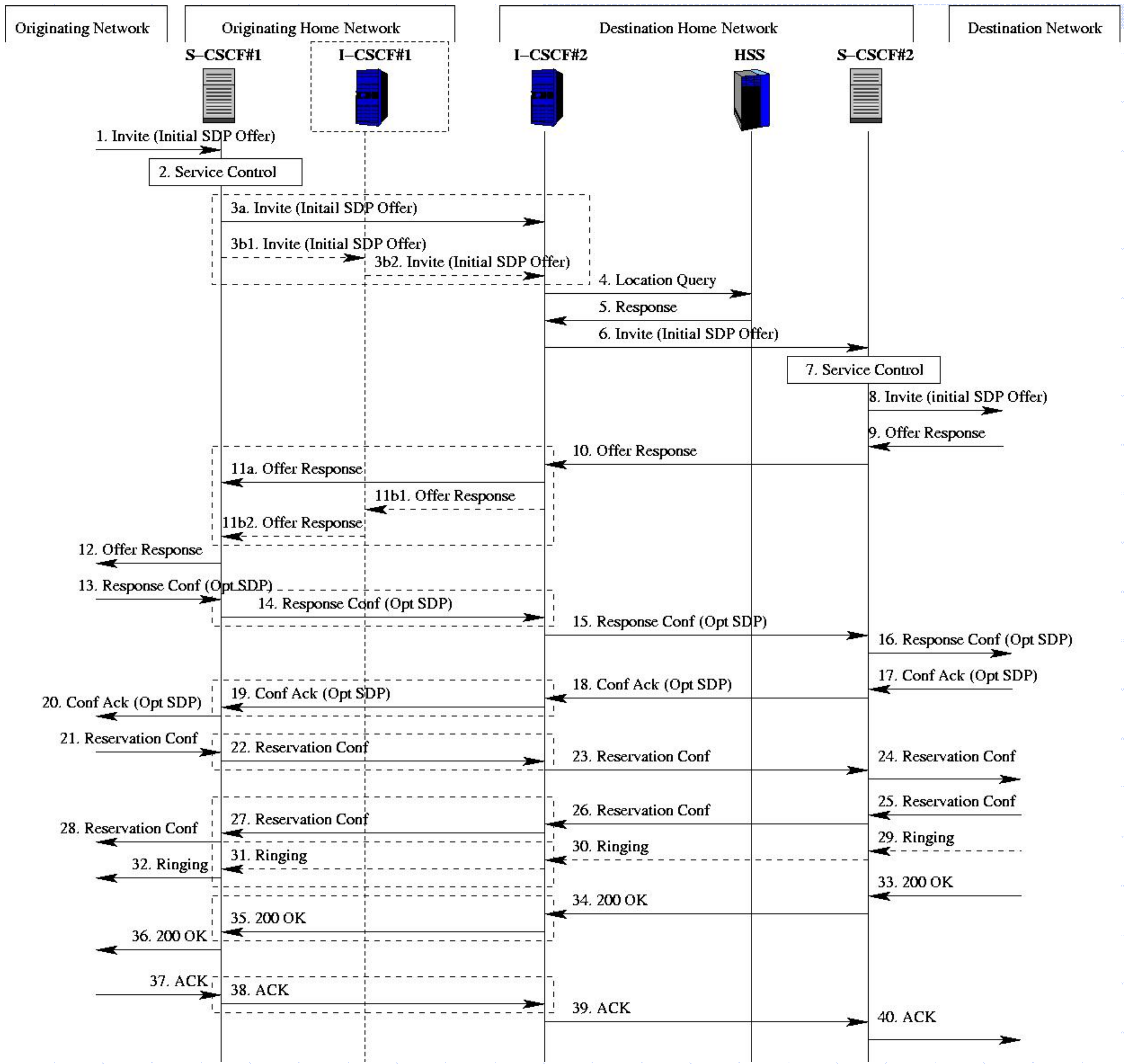


Fig. 3.16 S-CSCF to S-CSCF signaling flow

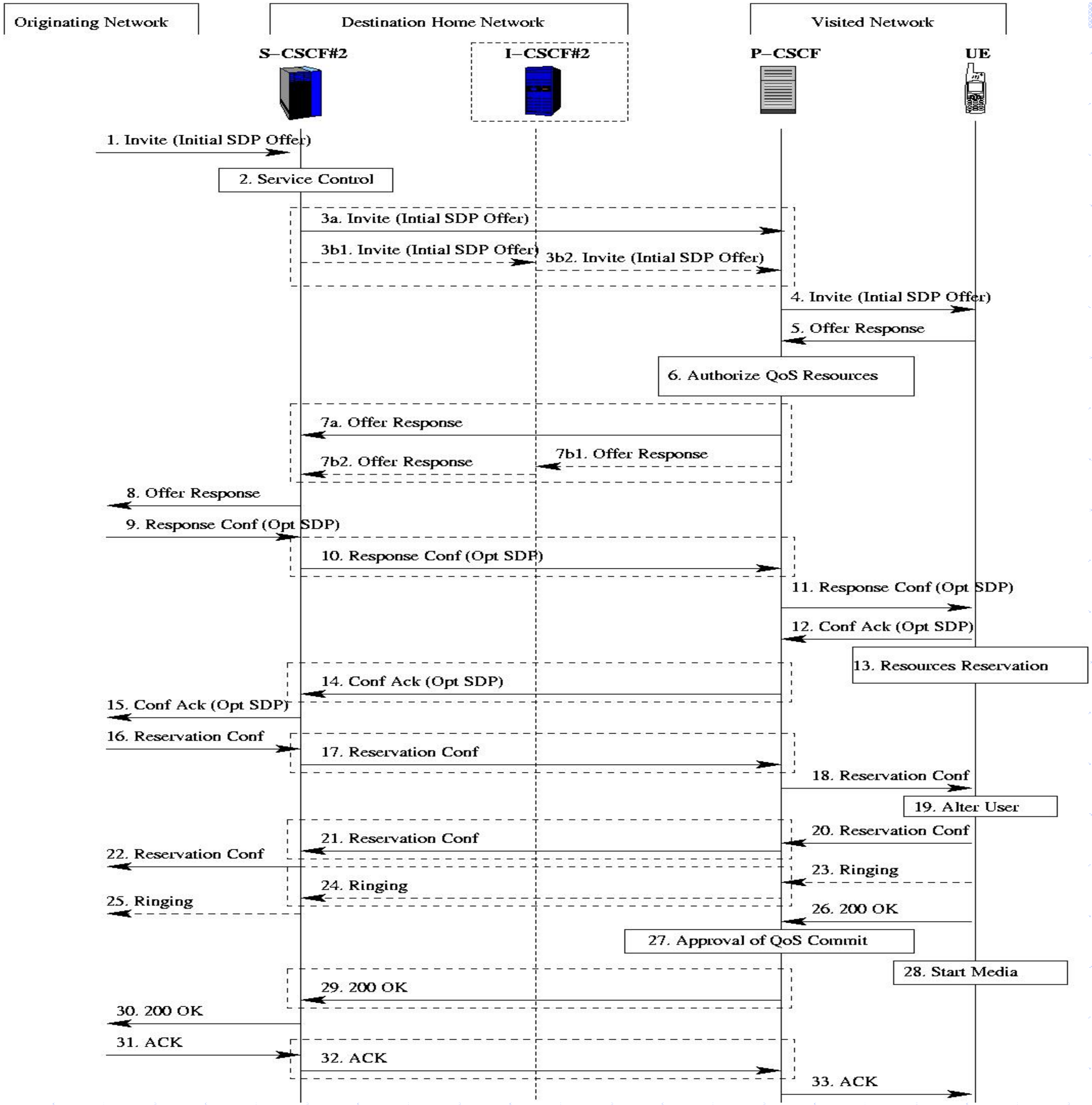
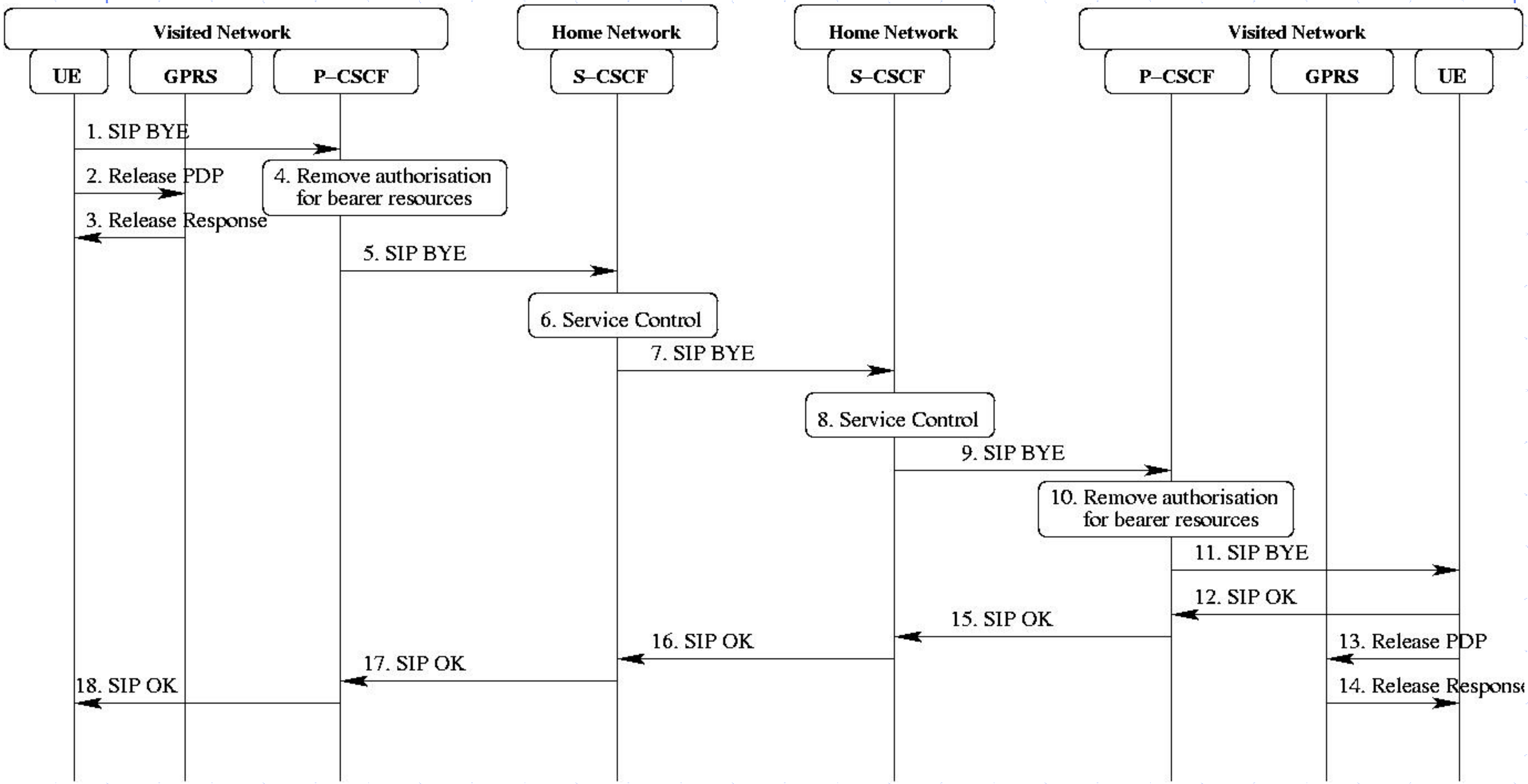


Fig. 3.17 Mobile termination flow

Fig. 3.18 Release flow: mobile initiated



3.3 3GPP2 IP MULTIMEDIA SUBSYSTEM (IMS)

- 3GPP2 IP Multimedia Domain (MMD): provide end-to-end IP connectivity, services, and features through the core network to subscribers
  - Packet Data Subsystem (PDS): support general packet data service
  - IP Multimedia Subsystem (IMS): provide multimedia session capabilities

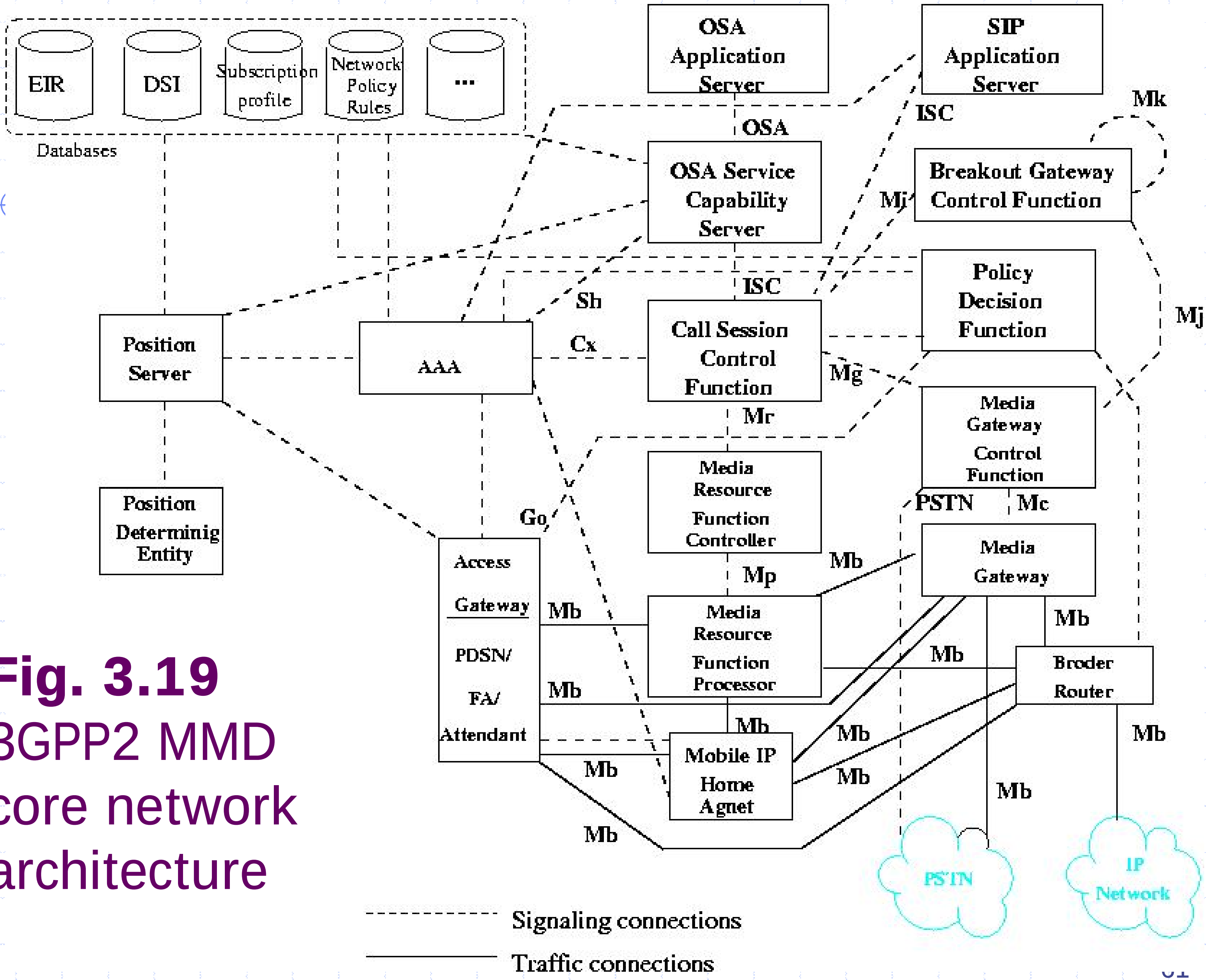


Fig. 3.19

3GPP2 MMD core network architecture

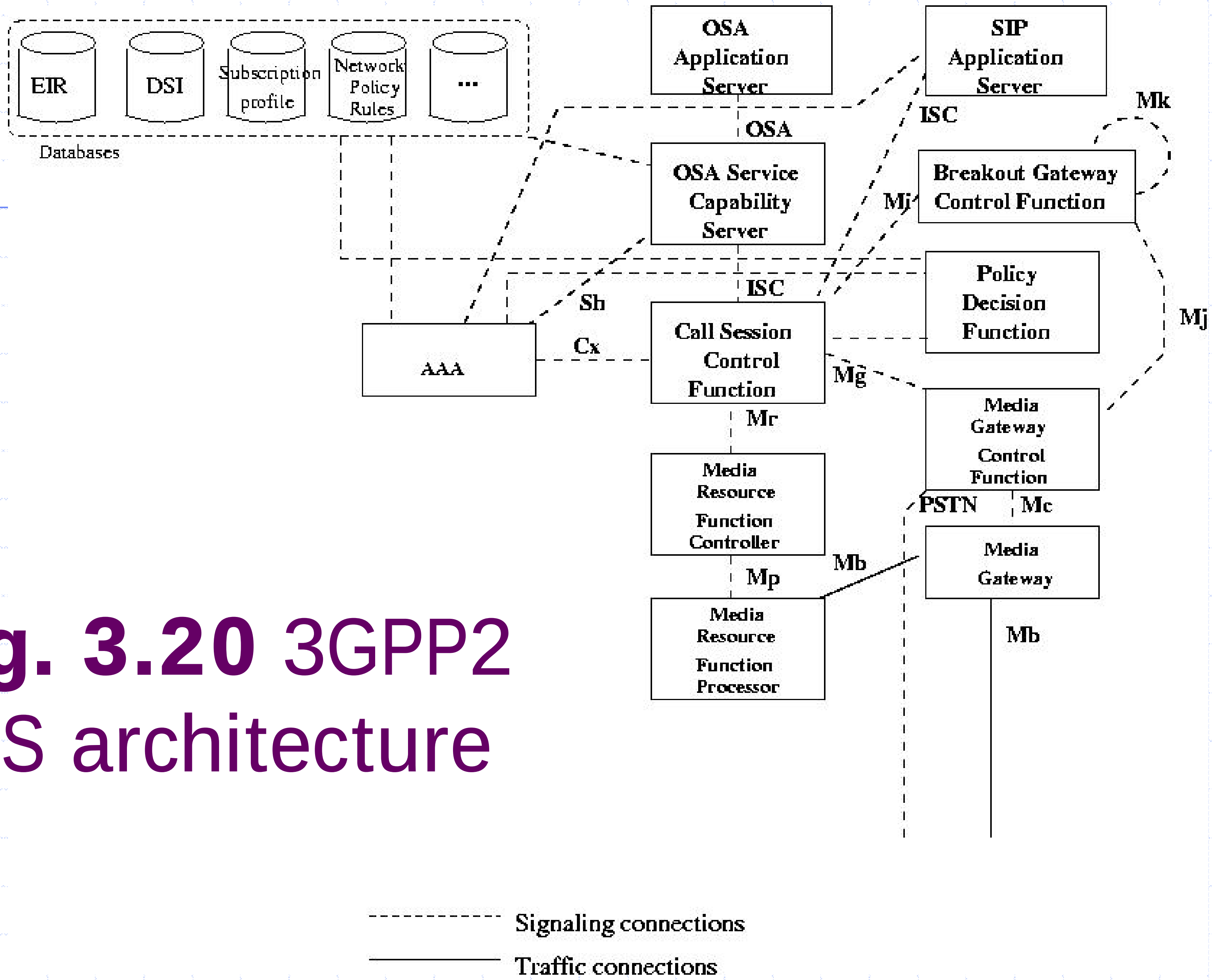


Fig. 3.20 3GPP2 IMS architecture

Fig. 3.21 3GPP2 IMS service platforms

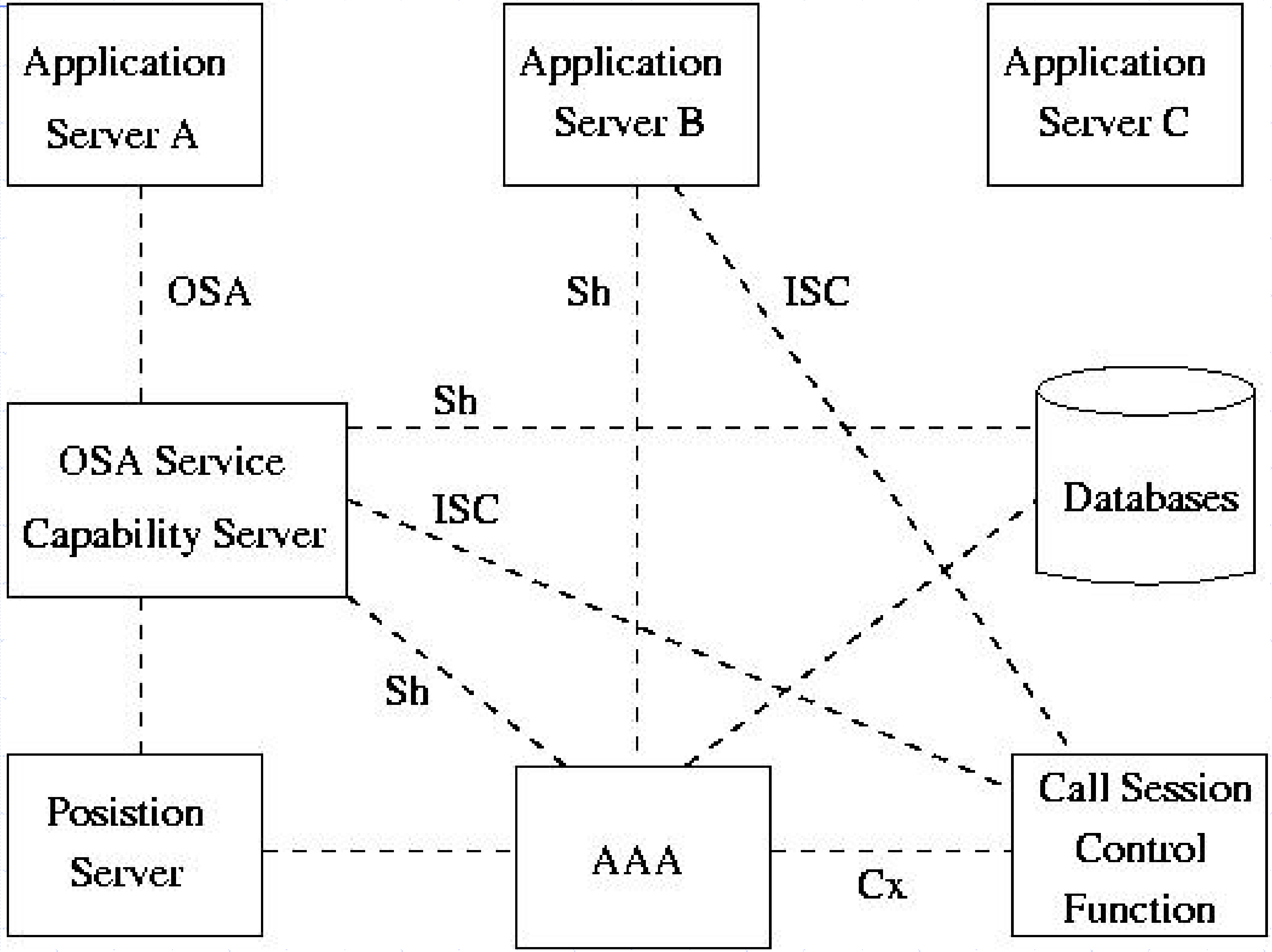


Fig. 3.22
3GPP2 IMS service control