SIP: Session Initiation Protocol

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Outline

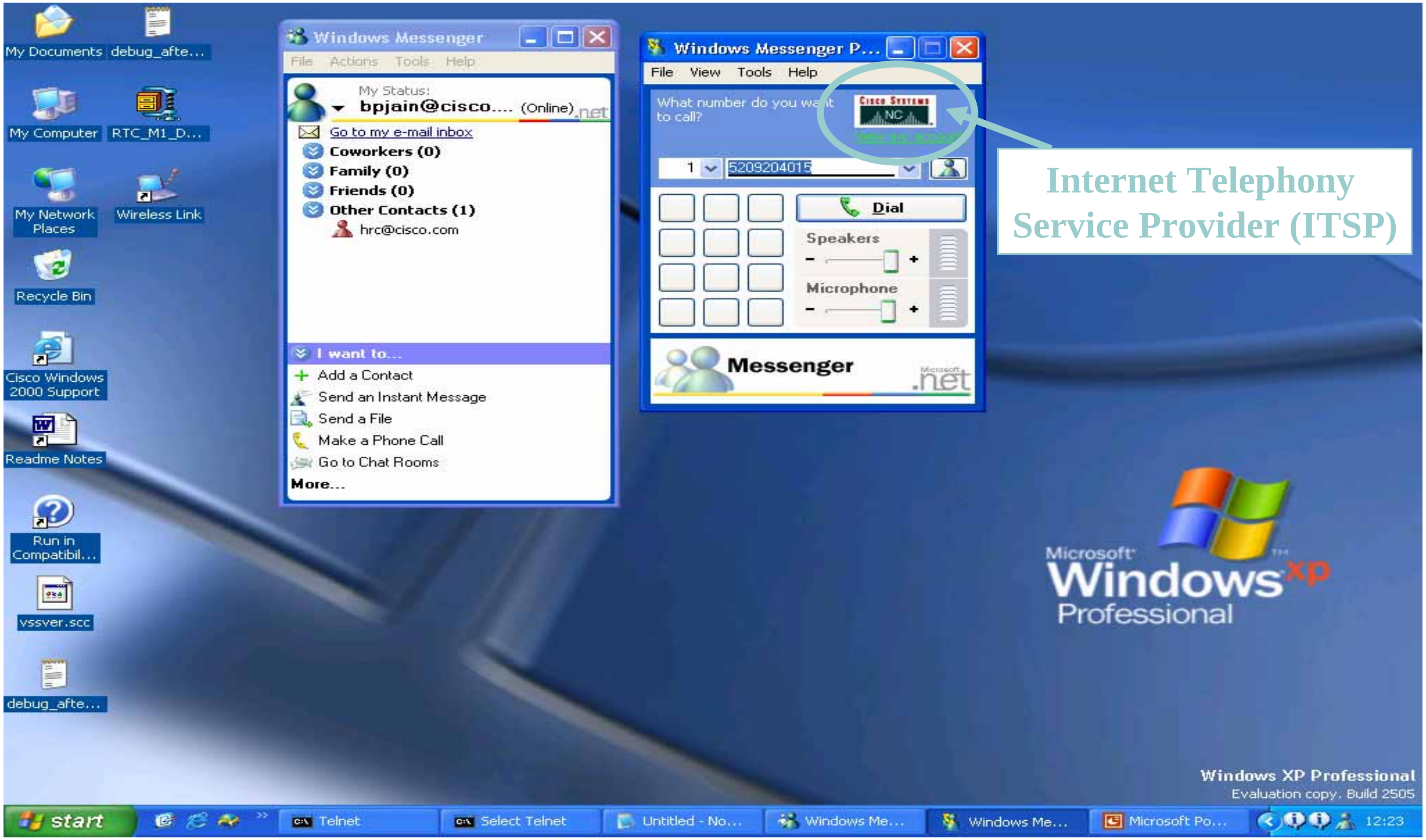
- Introduction to SIP
- SIP Architecture
- Mobility Management
- SIP and 3G Networks

Session Initiation Protocol (SIP)

Session Initiation Protocol

- SIP is originally proposed by Columbia University and is specified by IETF.
- SIP is an end-to-end application-layer protocol
  - Establish, modify and terminate interactive multimedia sessions, e.g., VoIP and video conference, between SIP-based users.
  - Signaling protocol.
  - Client-Server framework.
- H.323 is a alternative signaling protocol to support VoIP.

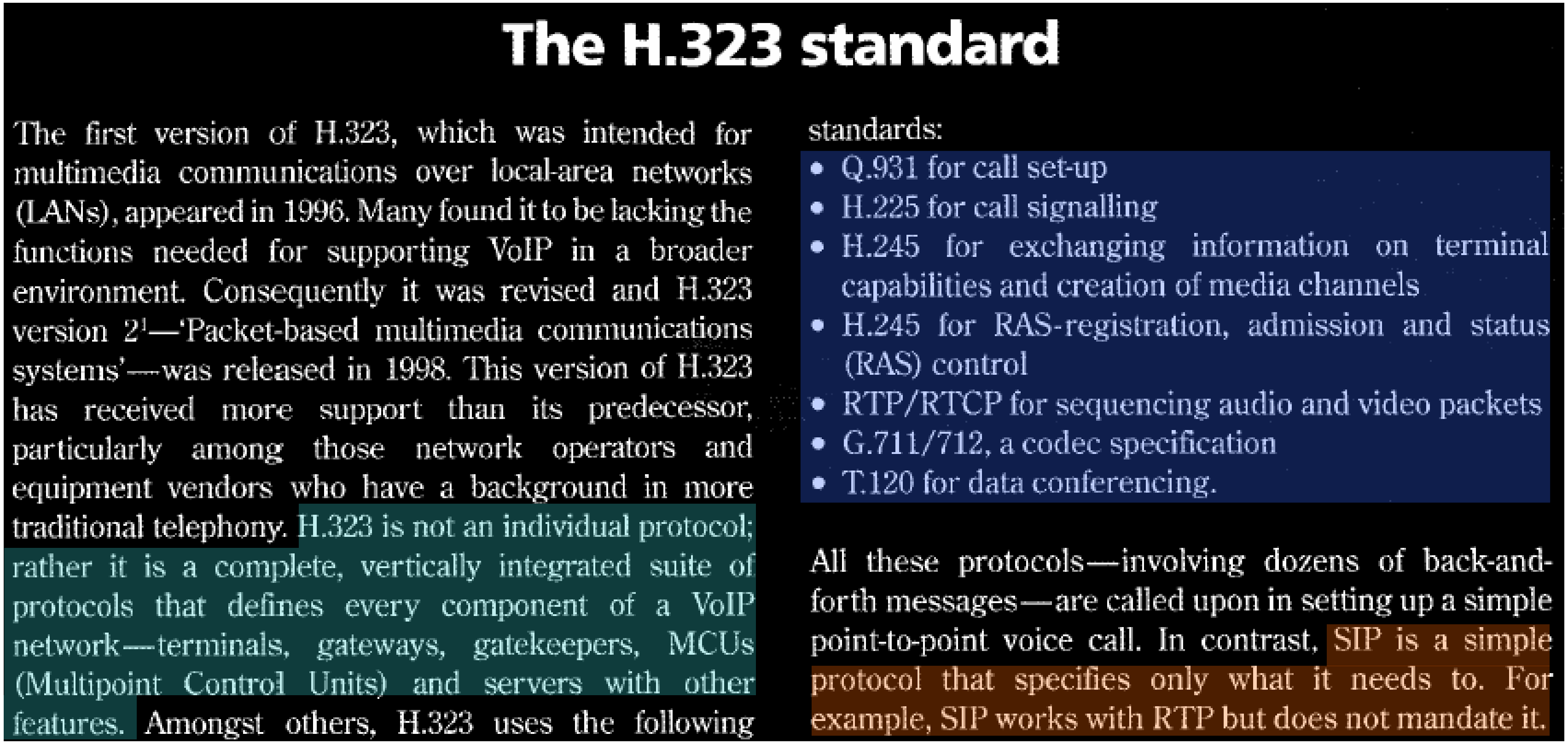
Microsoft Voice .NET Services



Features of SIP

- Text-based
  - Easy implementation in Java or Perl
    - JSIP open source library
  - Easy debugging
  - Flexible and extensible
- Less signaling comparing to H.323
  - QoS
- Transport-layer independence
  - UDP is commonly used.
- Forking a call request
  - Call forwarding
  - Parallel rings at different places

H.323


The H.323 standard

The first version of H.323, which was intended for multimedia communications over local-area networks (LANs), appeared in 1996. Many found it to be lacking the functions needed for supporting VoIP in a broader environment. Consequently it was revised and H.323 version 2[1]—'Packet-based multimedia communications systems'—was released in 1998. This version of H.323 has received more support than its predecessor, particularly among those network operators and equipment vendors who have a background in more traditional telephony. H.323 is not an individual protocol; rather it is a complete, vertically integrated suite of protocols that defines every component of a VoIP network—terminals, gateways, gatekeepers, MCUs (Multipoint Control Units) and servers with other features. Amongst others, H.323 uses the following standards:

- Q.931 for call set-up
- H.225 for call signalling
- H.245 for exchanging information on terminal capabilities and creation of media channels
- H.245 for RAS-registration, admission and status (RAS) control
- RTP/RTCP for sequencing audio and video packets
- G.711/712, a codec specification
- T.120 for data conferencing.

All these protocols—involving dozens of back-and-forth messages—are called upon in setting up a simple point-to-point voice call. In contrast, SIP is a simple protocol that specifies only what it needs to. For example, SIP works with RTP but does not mandate it.

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Four SIP Logical Entities

- User agent
- Proxy Server
- Registrar
- Redirect Server

User Agent

- User applications
- Both software and hardware

Type of SIP Servers

- Proxy Server
  - Application layer router used to relay SIP messages.
- Registrar
  - Accept registration request from user agent.
- Redirect Server
  - Redirects caller to other servers.

Typically, "SIP server" implements the functionality of Proxy, Registrar and Redirect Servers.

SIP Addressing

- SIP give you a globally reachable address.
  - Email-like address.
    - sip: leonard@a.ntu.edu.tw
    - sip: 82828888@a.ntu.edu.tw
- User agents bind this address to Registrar by using SIP REGISTER message.
- Each user agent communicates with one another by using this address.

SIP messages

Generic-message = start-line
*message-header
CRLF
[message-body]
start-line = Request-Line | Status-Line

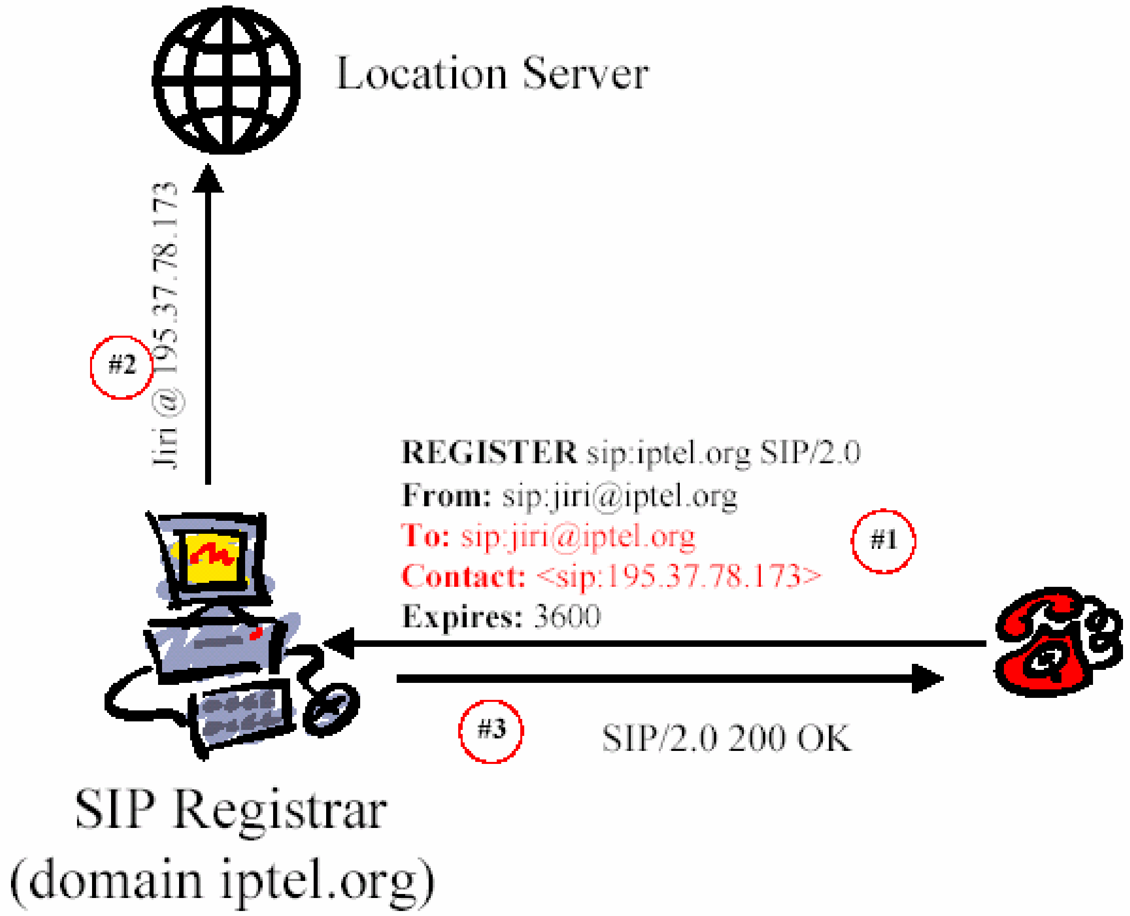
Request Line

SIP message	Description
INVITE	Invites a user to a call
ACK	Used to facilitate reliable message exchange for INVITEs
OPTIONS	Solicits information about a server's capabilities
BYE	Terminates a connection between users or declines a call
CANCEL	Terminates a request, or search, for a user
REGISTER	Registers a user's current location
INFO	Used for mid-session signalling

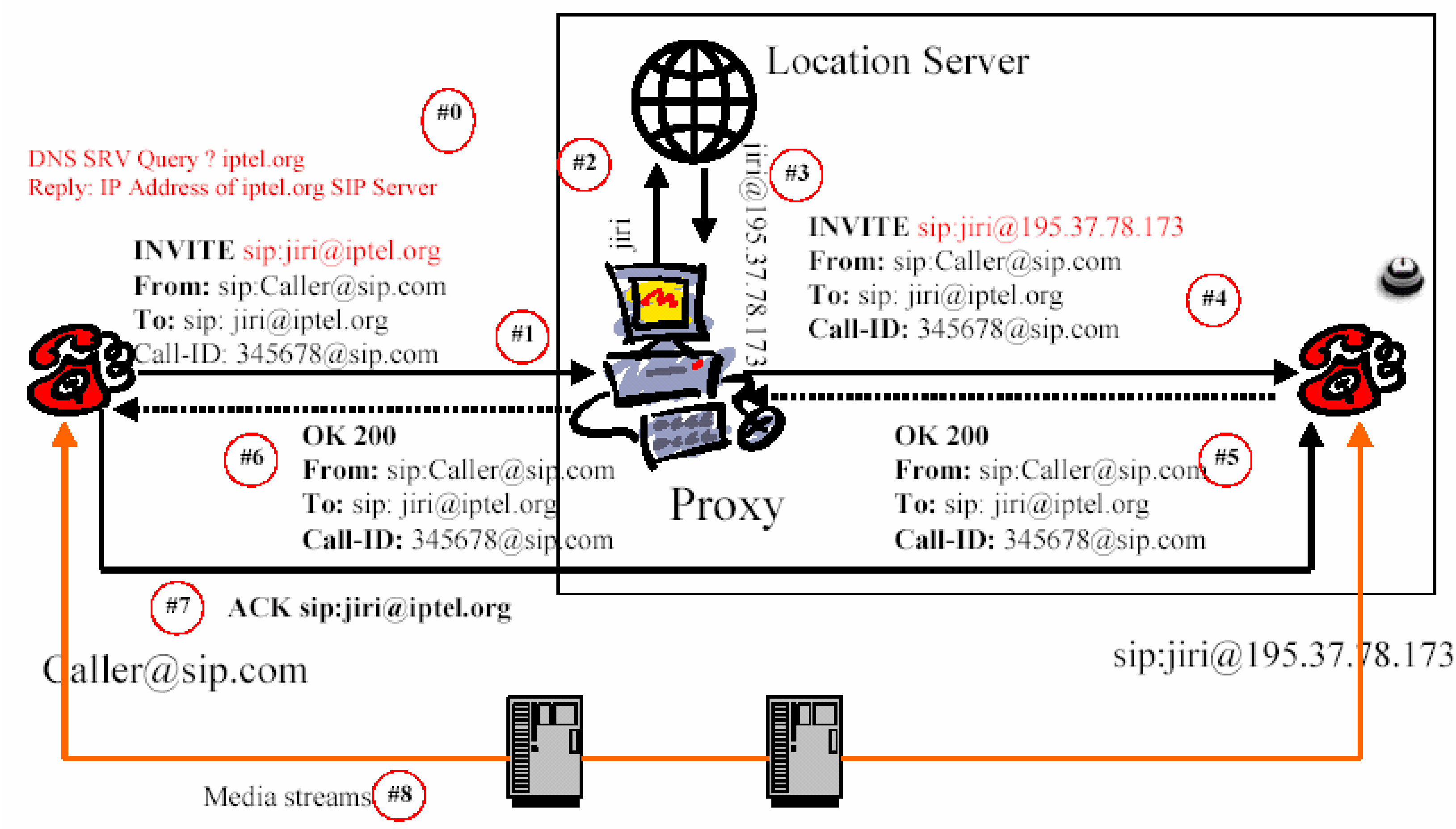
Class	Description	Example
1xx	Informational: request received, continuing to process the request	100 Trying, 180 Ringing
2xx	Successful: the action was successfully received, understood and accepted	200 OK
3xx	Redirection: further action needs to be taken in order to complete the request	302 Moved Temporarily
4xx	Client Error: the request contains bad syntax or cannot be fulfilled at this server	404 Not Found
5xx	Server Error: the server failed to fulfil an apparently valid request	501 Not Implemented
6xx	Global Failure: the request cannot be fulfilled at any server	603 Decline

Status Line

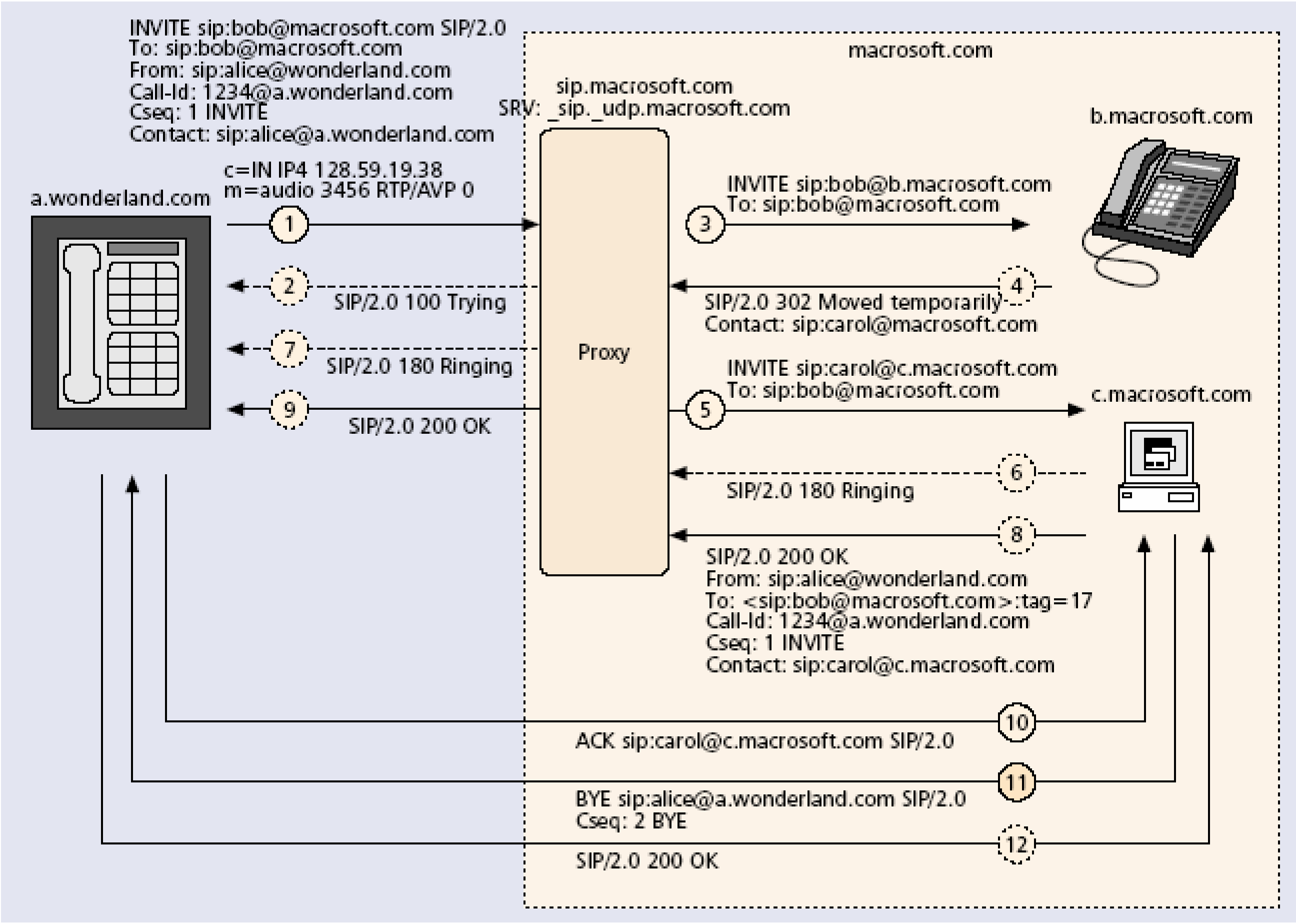
Example: SIP Registration



Example: Session Establishment



Example: Session Forwarding



Session Description Protocol (SDP)

- The message body of SIP
- SDP is used to describe a multimedia session

```
u = http://www.ietf.org
e = g.bell@bell-telephone.com
c = IN IP4 132.151.1.19
m = audio 3456 RTP/AVP 96
a = rtpmap:96 VDVI/8000/1
m = video 3458 RTP/AVP 31
m = application 32416 udp wb
a = orient:portrait
```

RTP, RTCP, and RTSP

- Real Time Transport Protocol (RTP)
  - Encode and decode media stream
  - Recover the possible loss and jitter
- Real Time Control Protocol (RTCP)
  - QoS feedback
  - ...
- Real Time Streaming Protocol (RTSP)
  - Control stored media
  - VCR remote control
  - Support play, record , pause, fast forward, and etc.

RTSP protocol session

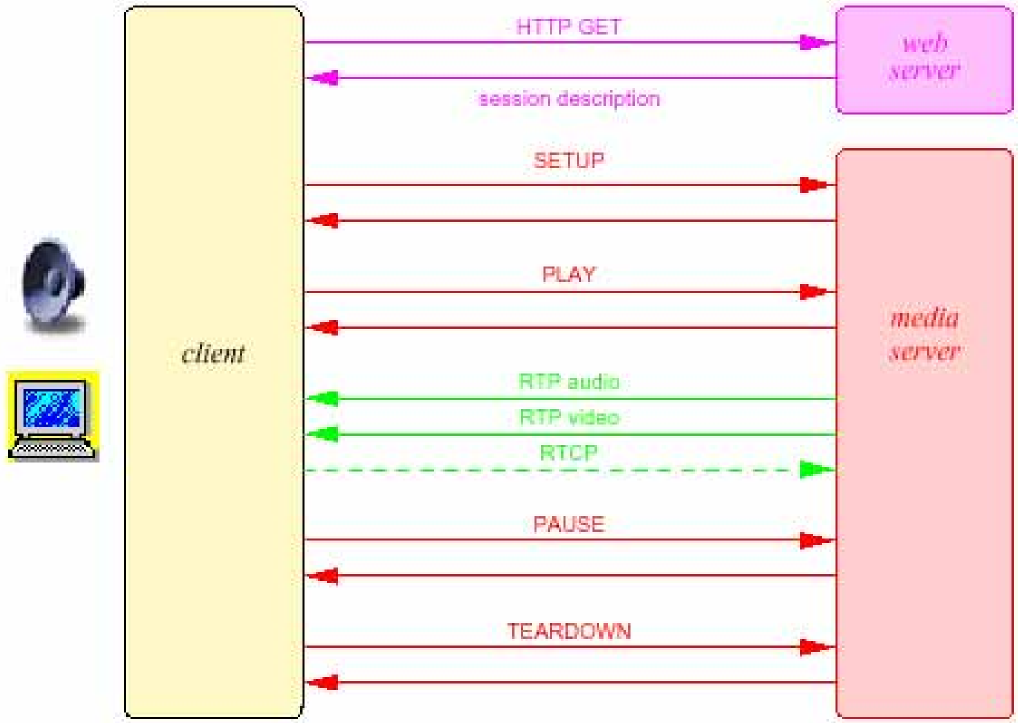
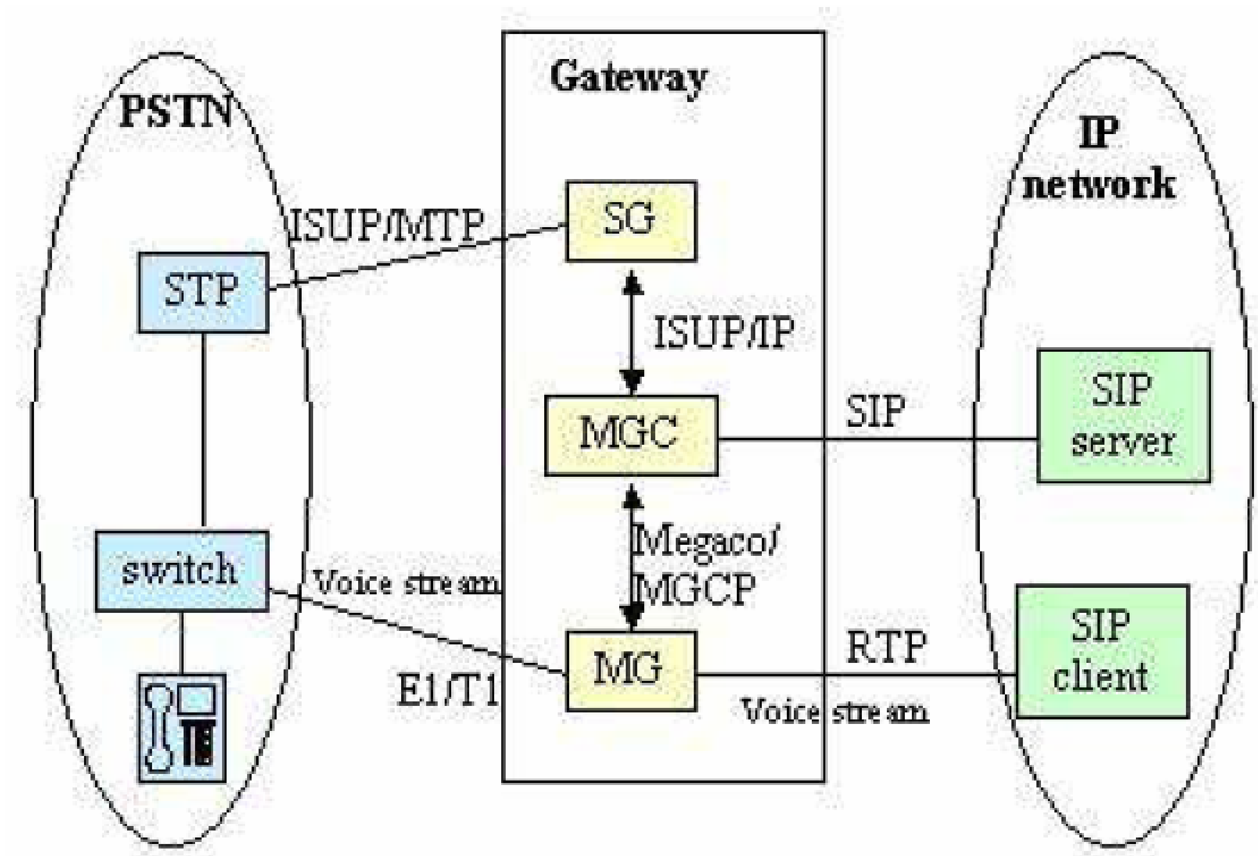


Figure 6: RTSP protocol session

SIP Interworking with the SS7



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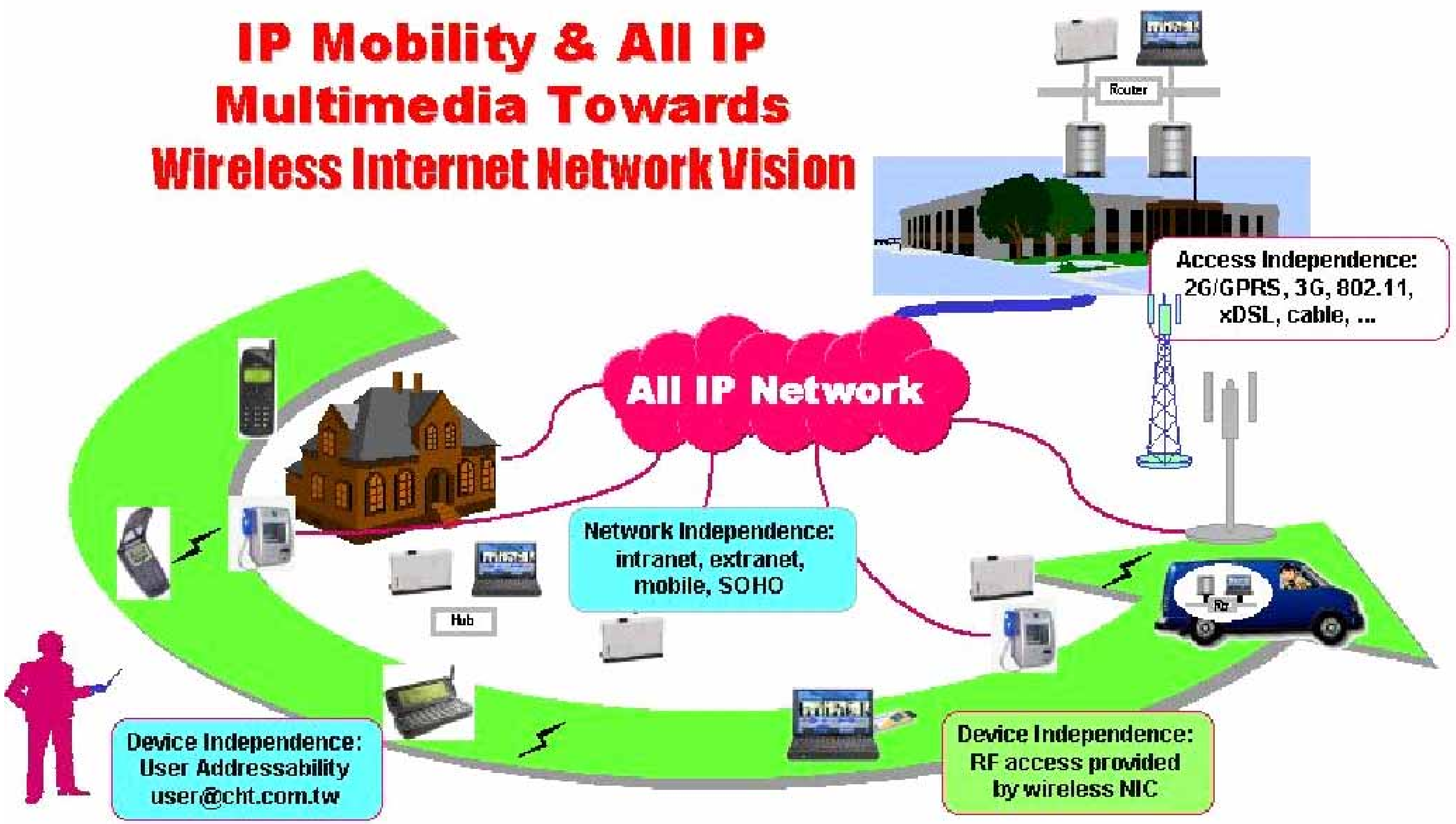
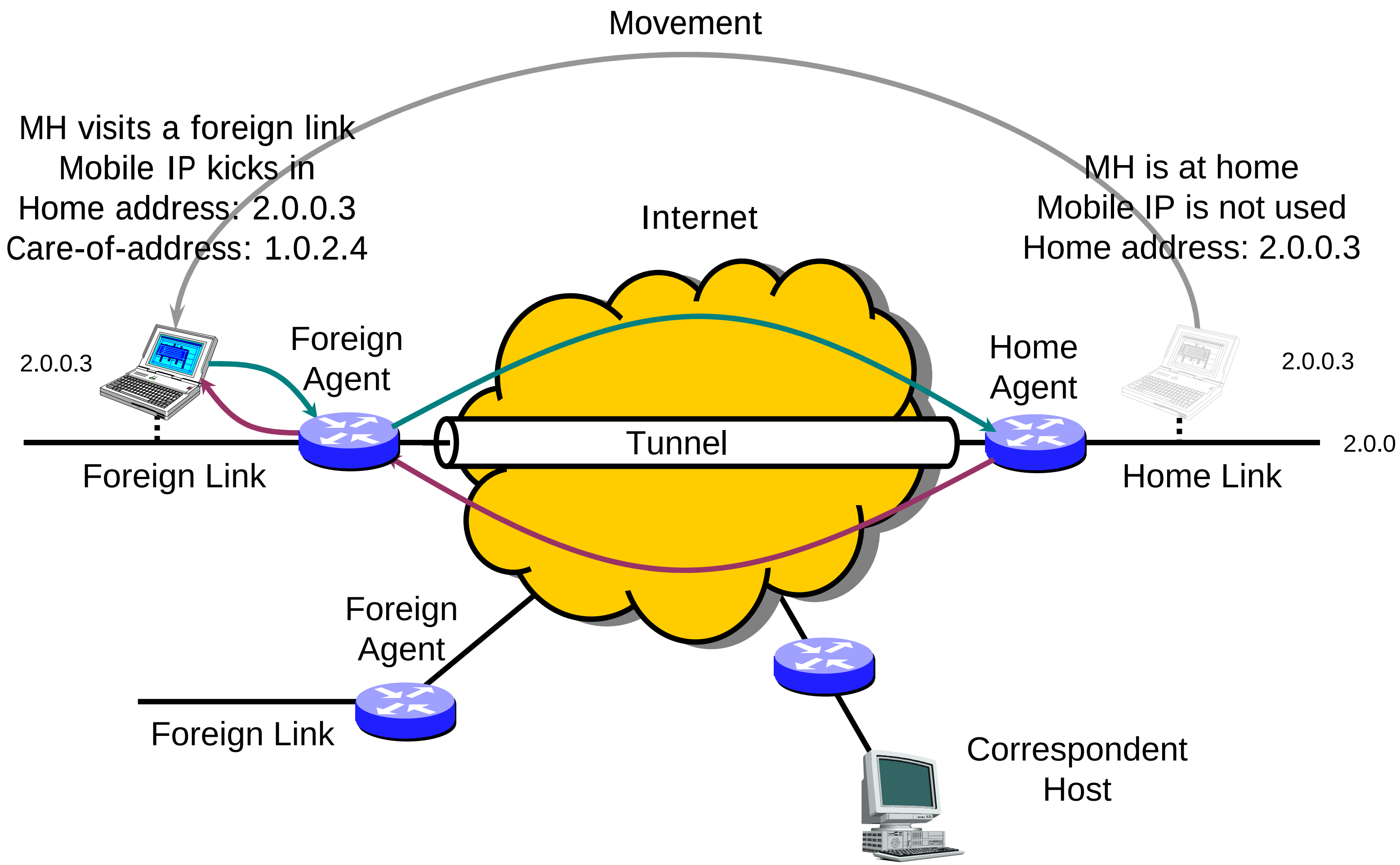
Wireless Technologies Convergence



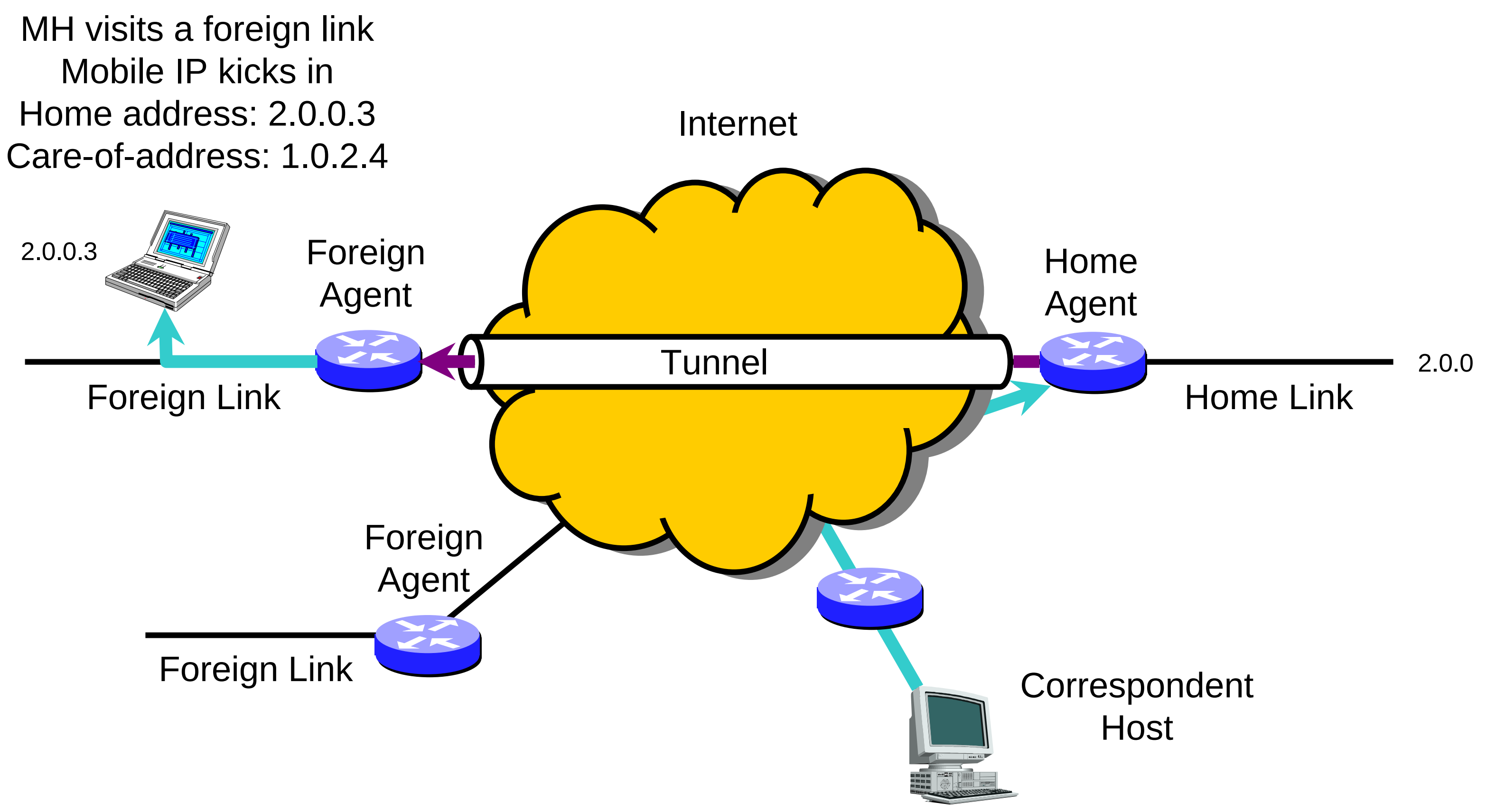
Mobility Management

- Mobility Classification
  - Roaming
  - Macro-mobility
    - Domain mobility
  - Micro-mobility
    - Subnet mobility
- Solutions
  - Network layer solution: Mobile IP
  - Application layer solution: SIP

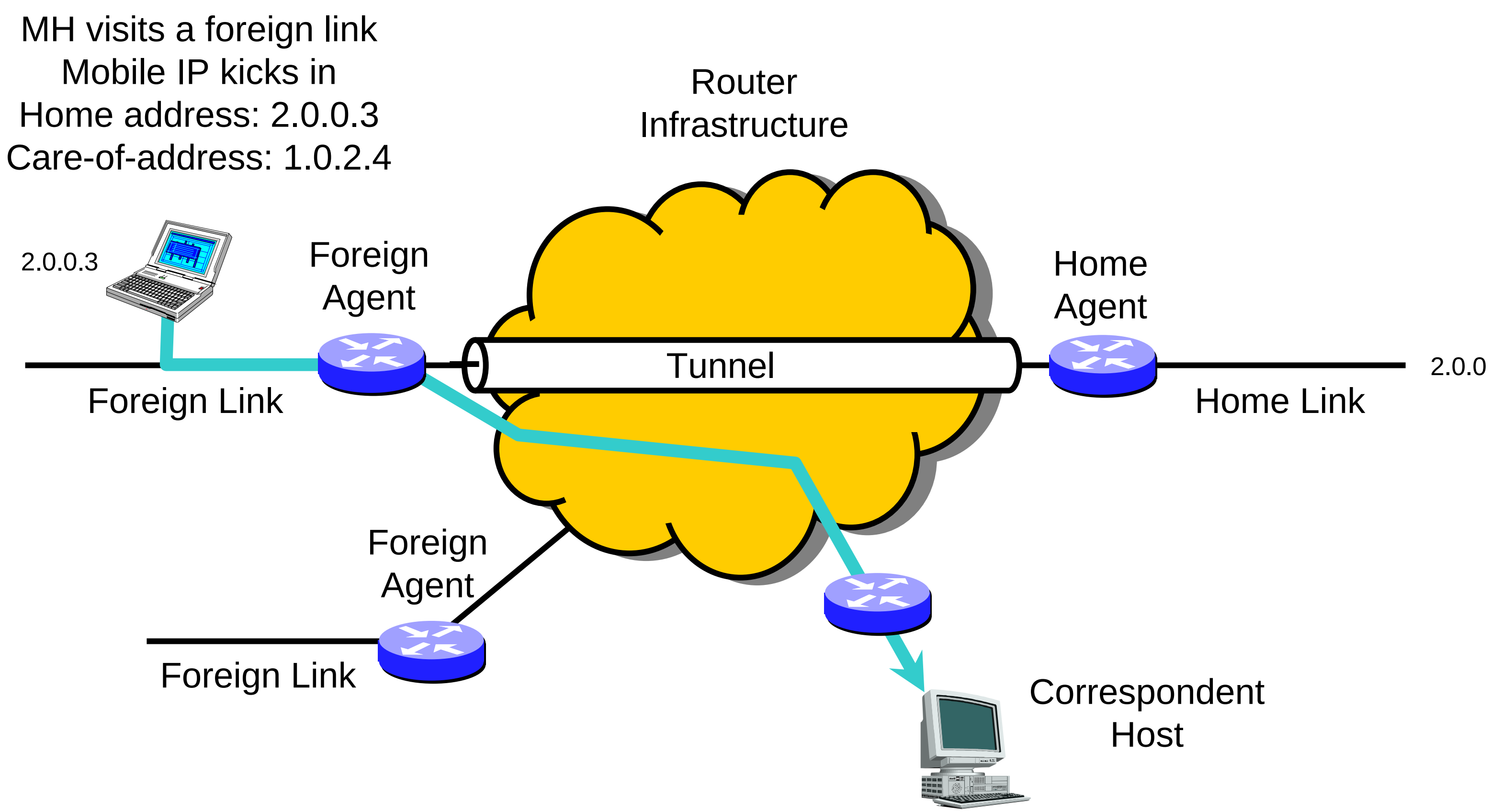
Mobile IPv4: Registration Example



Mobile IPv4: CH-to-MH Routing Example



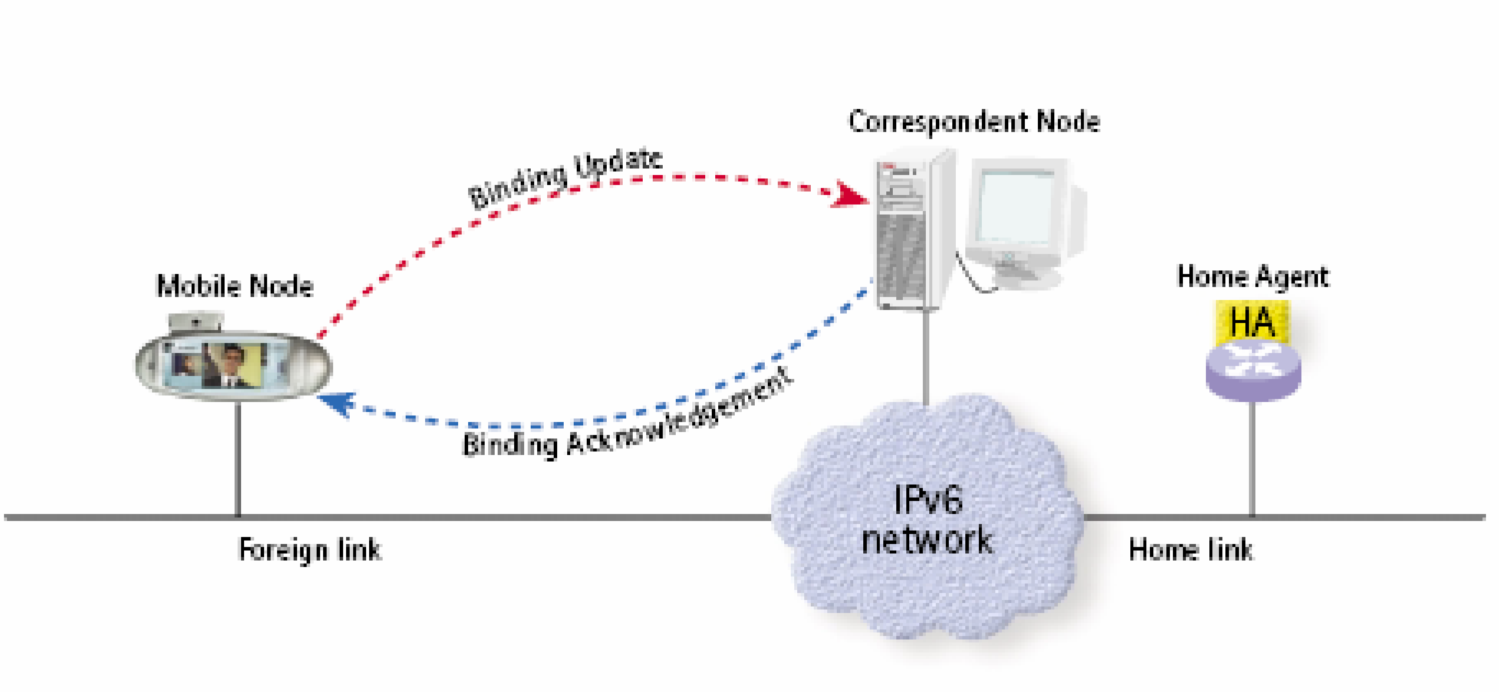
Mobile IPv4: MH-to-CH Routing Example



Mobile IPv4

- Triangle route problem
- Micro-mobility improvement
  - Cellular IP, Campbell in Column University.
  - Regional Registration, Perkins, Nokia Center.
  - ...

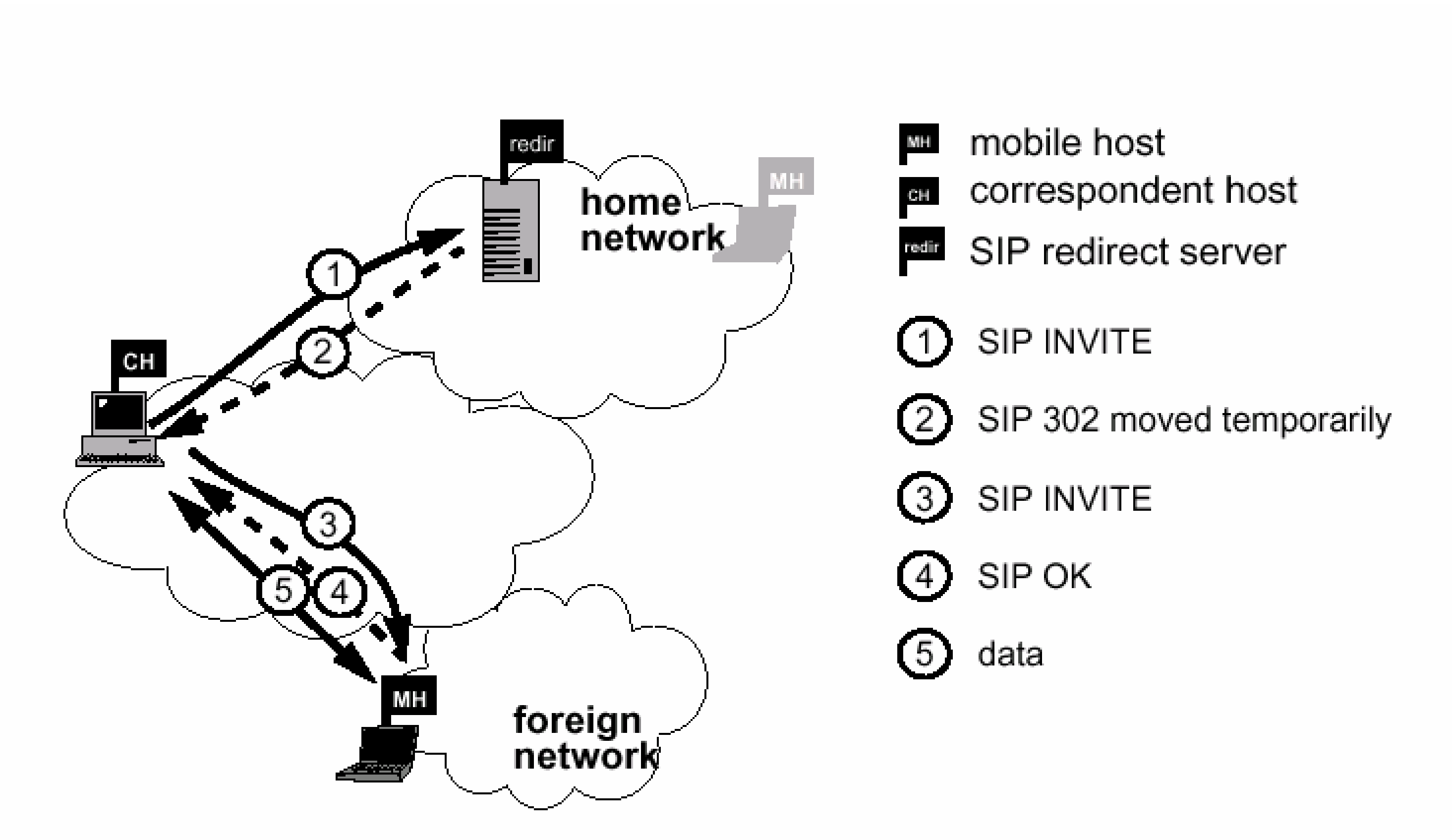
Mobile IPv6: Binding Update



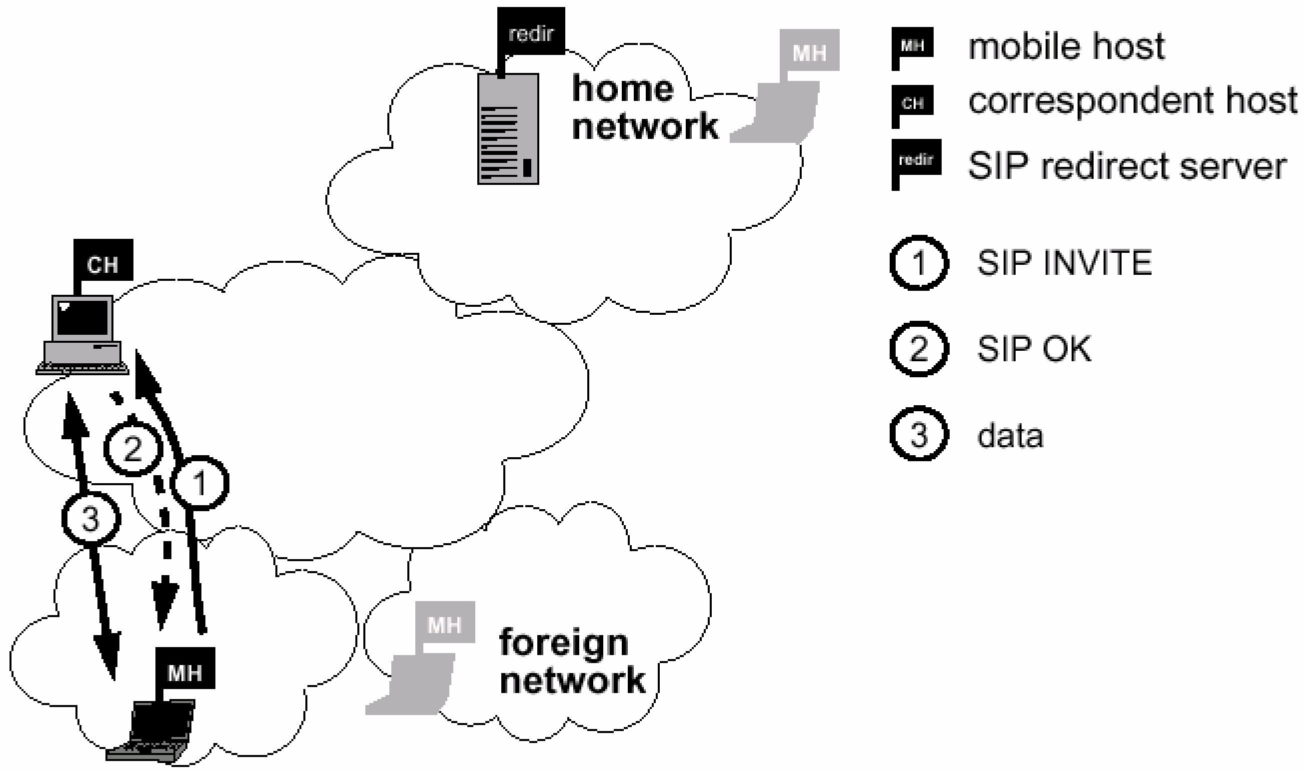
Application Layer Mobility Using SIP

- Terminal Mobility
- Session Mobility

Terminal Mobility

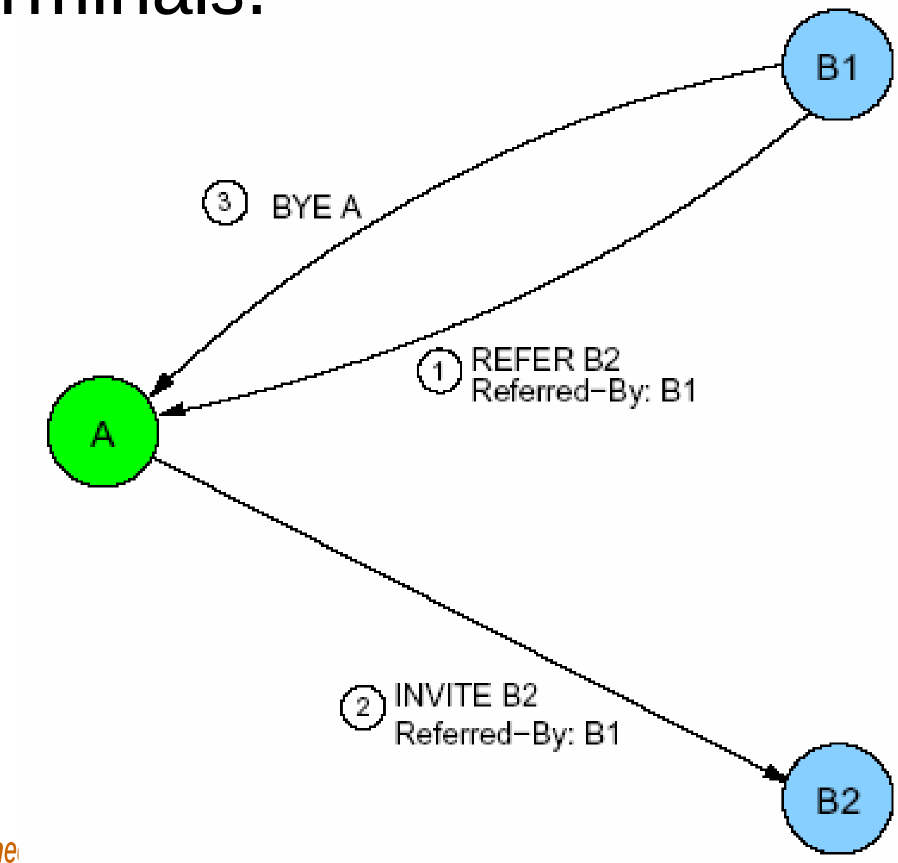


Terminal Mobility

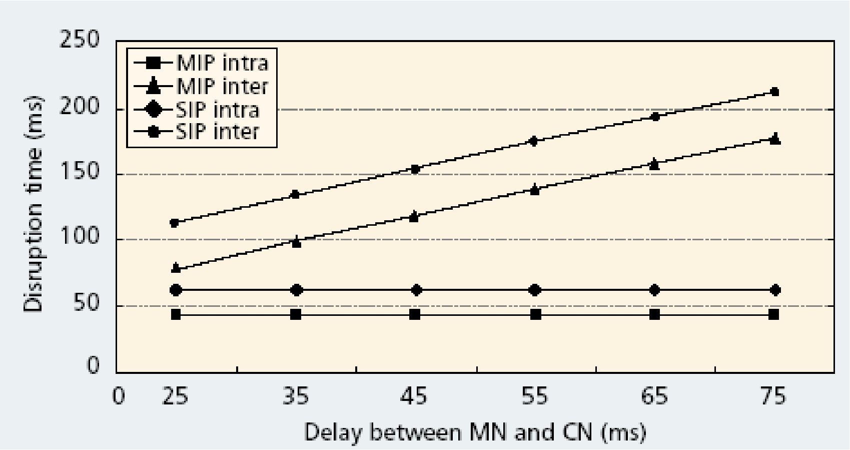


Session Mobility

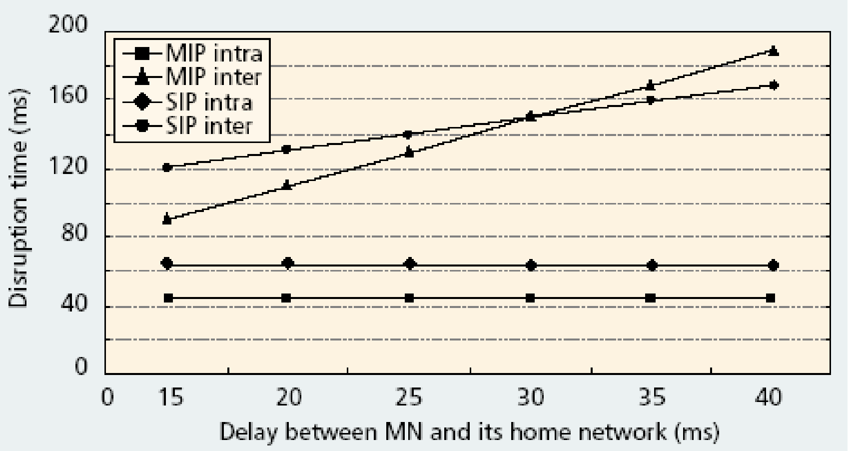
- Allow a user to maintain a media session even while changing terminals.



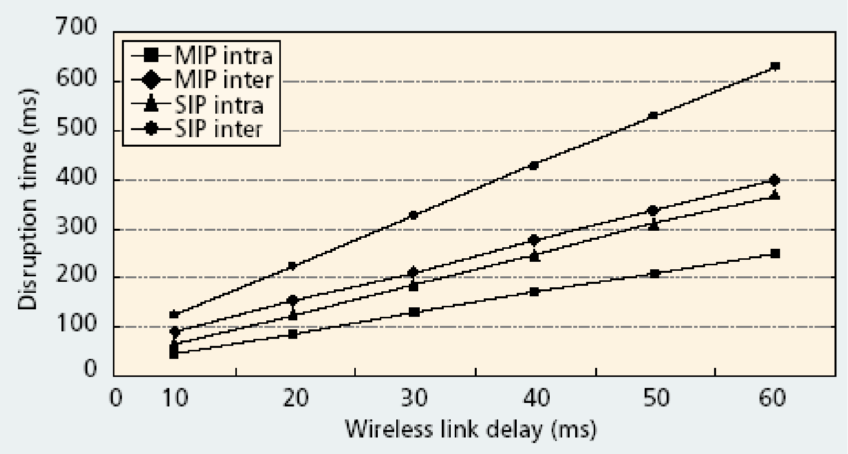
Comparison



MIP intra
MIP inter
SIP intra
SIP inter
Disruption time (ms)
Delay between MN and CN (ms)



MIP intra
MIP inter
SIP intra
SIP inter
Disruption time (ms)
Delay between MN and its home network (ms)



MIP intra
MIP inter
SIP intra
SIP inter
Disruption time (ms)
Wireless link delay (ms)

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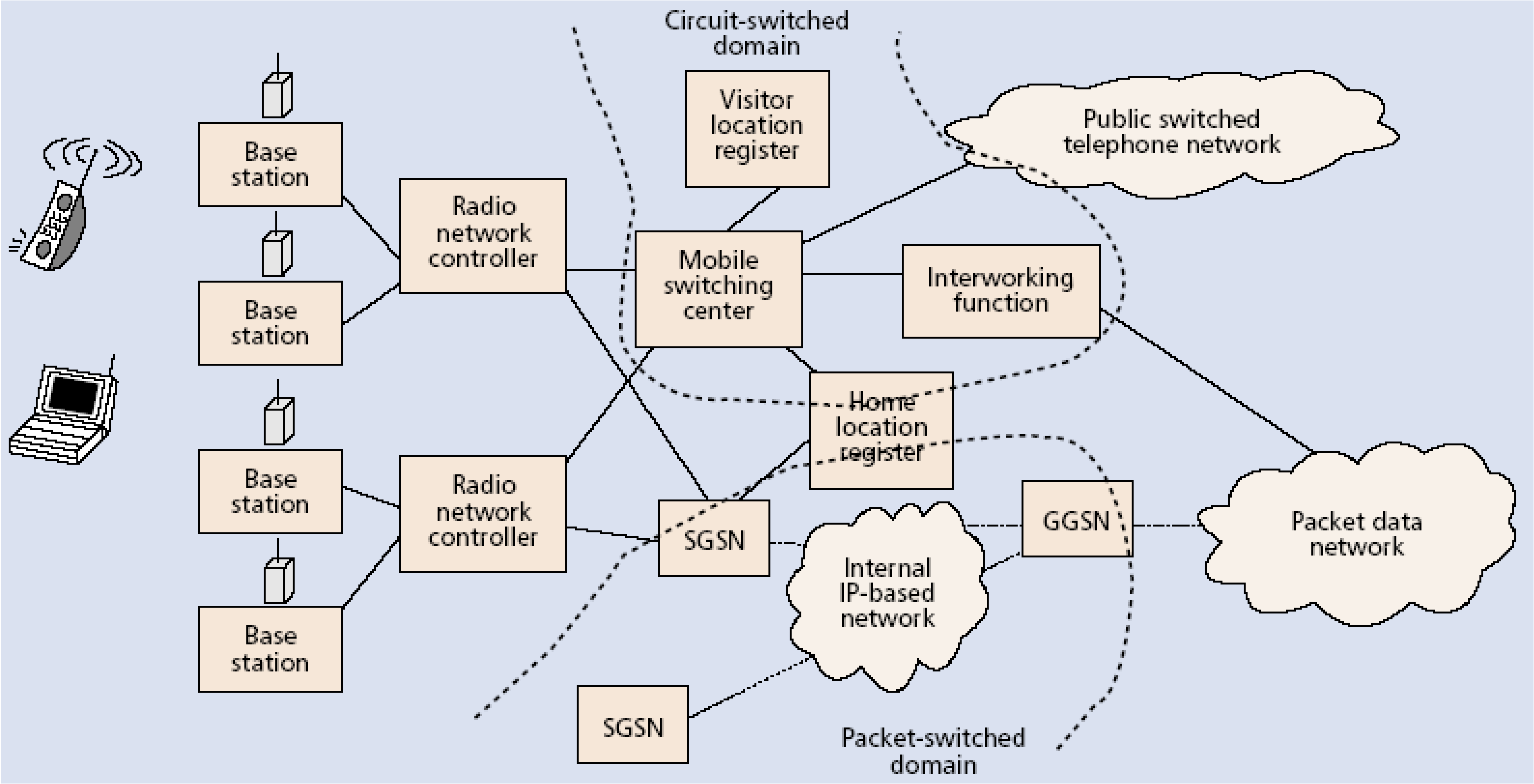
3G UMTS


Circuit-switched domain
Visitor location register
Base station
Radio network controller
Mobile switching center
Public switched telephone network
Interworking function
Base station
Home location register
Base station
Radio network controller
SGSN
GGSN
Packet data network
Base station
Internal IP-based network
SGSN
Packet-switched domain

Pure IP connectivity vs. Dedicated Multimedia subsystem

- Some mechanisms should be defined in 3G to support multimedia session transfers?
- Market Perspective
  - Subscriber perspective
  - Network operator perspective
  - Third-party service provider perspective

Subscriber Perspective

- Advantages
  - It is free and flexible to choose applications.
  - Reuse application in wired-networks
- Disadvantage
  - Trouble to choose the application and service provider.
  - The demand of service package and one bill.
  - Some application may lose QoS guarantee.

Network Operator Perspective

- Advantages
  - Operators may not have experience in IP multimedia applications. They only focus in the IP connectivity.
- Disadvantages
  - Circuit-switch revenue will be decayed.
  - Loss possible revenue for paving basic IP multimedia application, e.g., VoIP.
  - Issue of customer dissatisfaction for IP multimedia applications.

Third-Party Service Provider Perspective

- Advantages
  - They don’t have to bother the peculiarity of wireless networks. They don not need extensive knowledge of wireless telecommunication networks and protocols.
- Disadvantages
  - They are unable to take advantage of the wireless network, e.g., user location information.

IP Multimedia Subsystem (IMS)

- Appear in Release 5 and beyond
- IMS comprises the network elements for control of multimedia sessions.
- Network operator provides both
  - IP connectivity
  - Multimedia session management

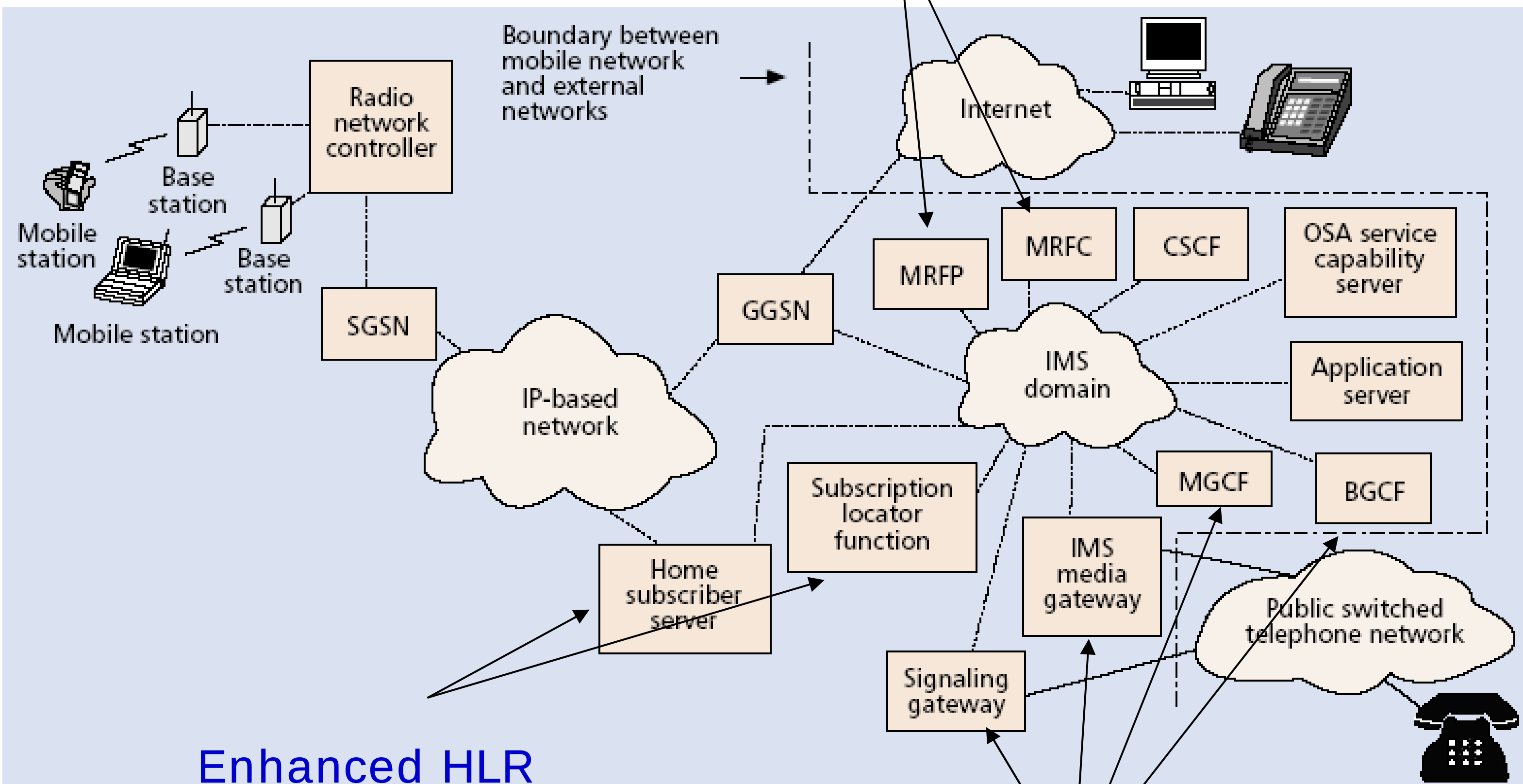
Basic Add-in Features

- Call State Control Function (CSCF)
  - Provisioning of call control for IP multimedia applications. P-CSCF, I-CSCF, S-CSCF.
- Open Service Access (OSA)
  - Third-party are expected to stimulate innovative application, taking advantage of knowing the capabilities provided by wireless network providers.

IP Multimedia Subsystem

Media Resource Function Processor

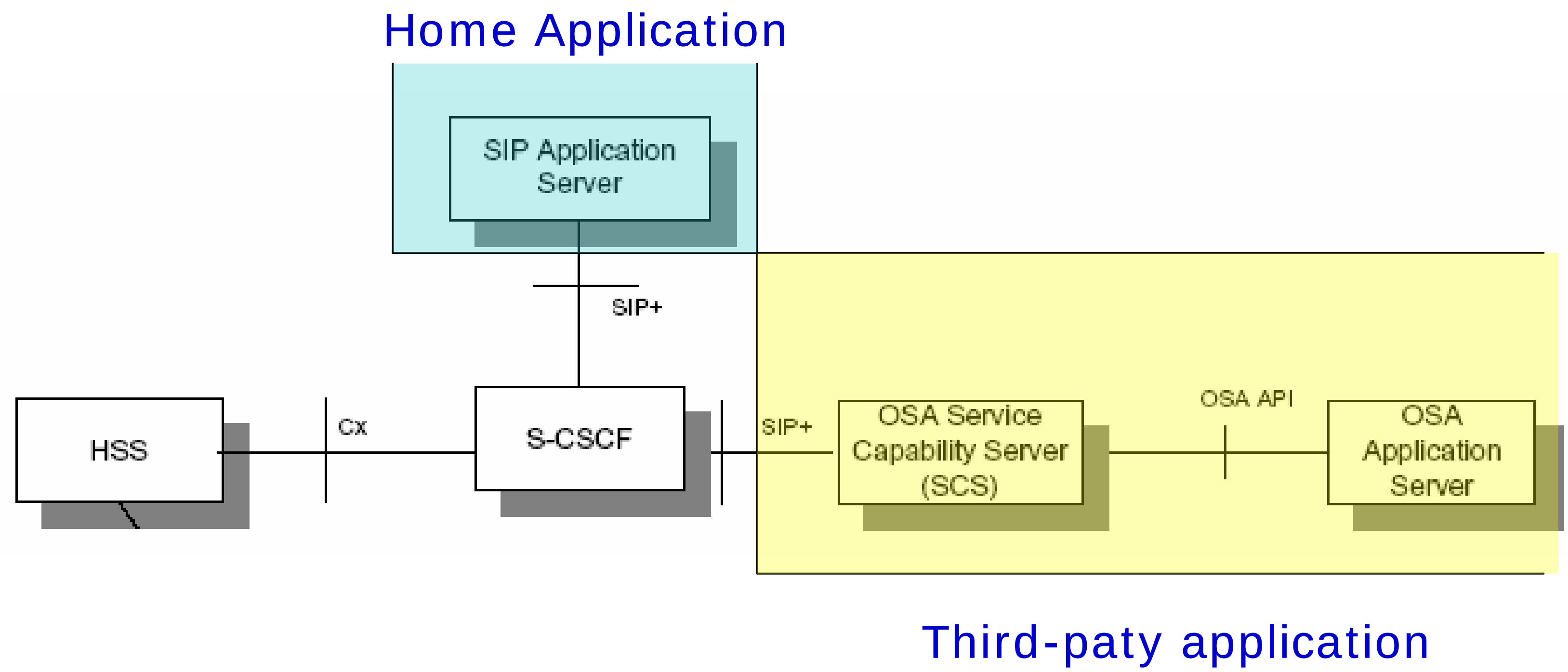
Media Resource Function Controller



Enhanced HLR

Perform internetworking related functions with PSTN

Serving-CSCF



S-CSCF

- Session control.
- Retrieve the information from HSS.
- Connect to Application Servers.
- Each user agent needs to attach a S-CSCF before setup a session.
- Analog to Registrar in SIP.

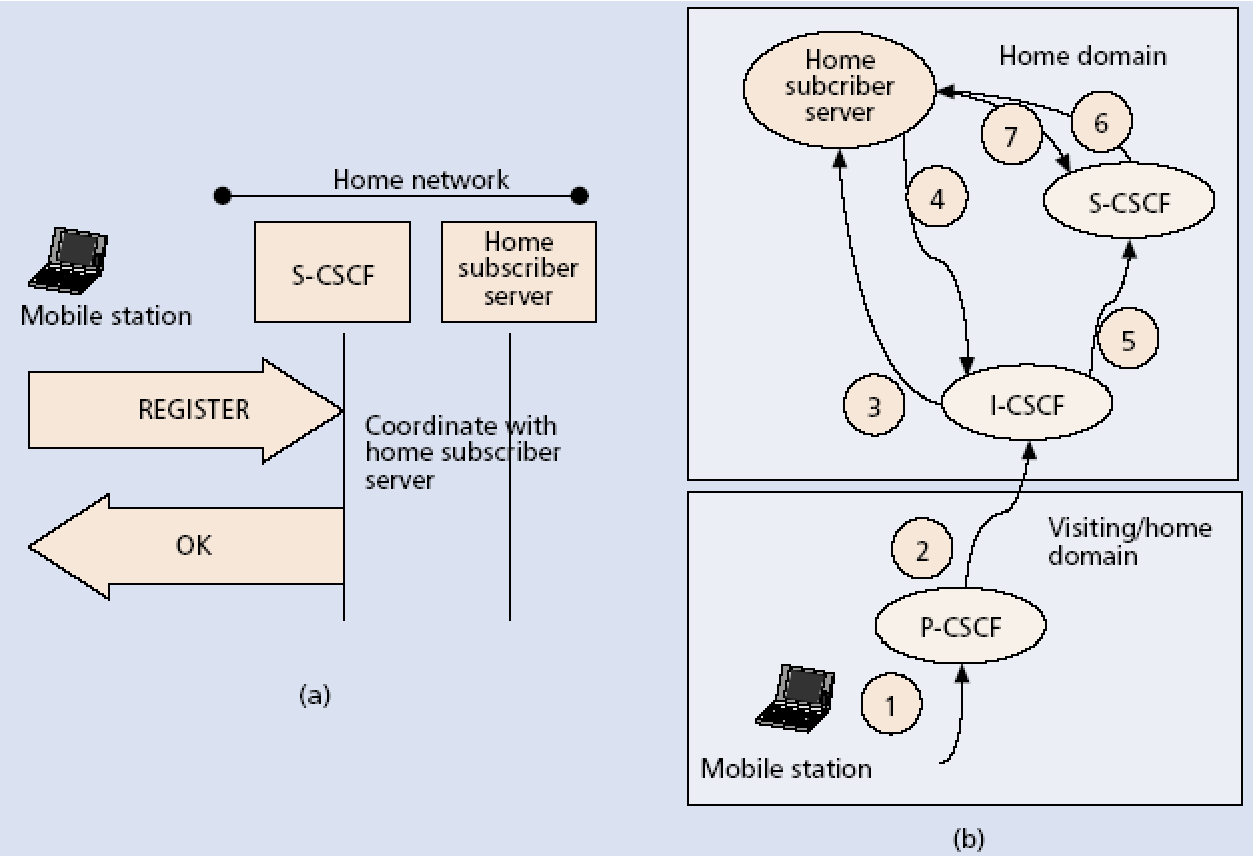
Proxy-CSCF (P-CSCF)

- The first contact point within the IMS.
- Mobile node communicate with S-CSCF via P-CSCF. Direct communication with S-CSCF is not allowed.
  - Integrity protection of SIP signaling.
  - Compression due to sparse wireless resource (Sigcom).
  - Inspect SIP signaling if the mobile node is in a visited network.

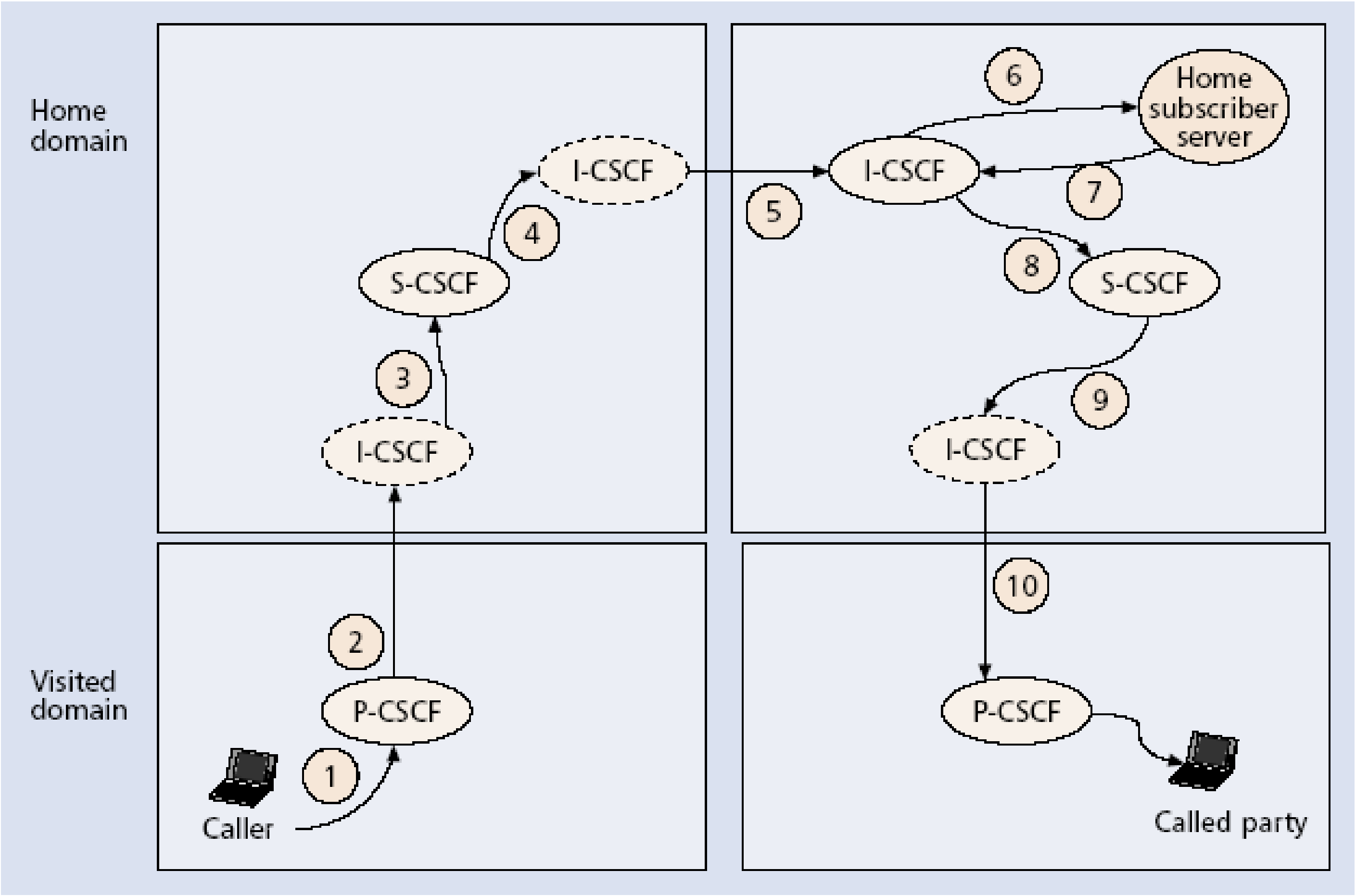
Interrogating-CSCF (I-CSCF)

- Entry Point in a network operator.
- Hide the configurations, topology and capacity from outside.
- Analog to Proxy and redirect servers in SIP.

Registration



Session Setup



Reference

Mobile IP: Charles E. Perkins http://people.nokia.net/~charliep

SIP: http://www.cs.columbia.edu/sip

IMS: 3GPP TS 23.228 v2.0 http://www.3gpp.org/ftp

Thanks !!