

# SIP: Session Initiation Protocol



Eric Wu  
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## Outline

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

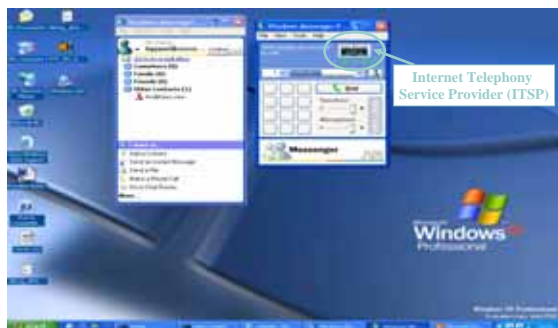
## Session Initiation Protocol (SIP)



## 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 an 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—“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/G.722, 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

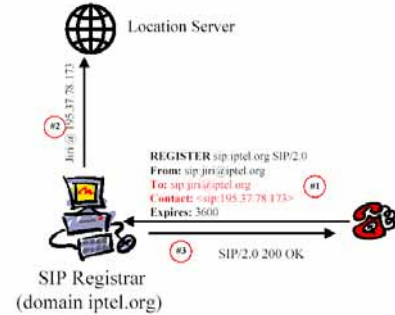
Generic message = start-line  
 \*message-header  
 CRLF  
 [message-body]  
 start-line = Request-Line | Status-Line

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

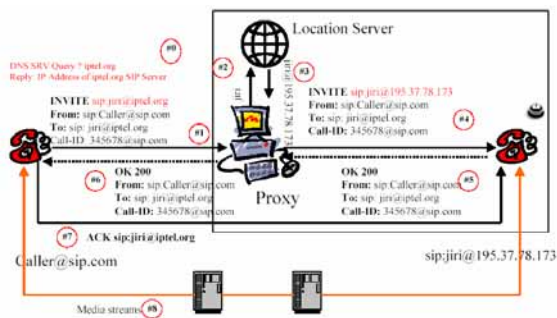
  

Class	Description	Example
1xx	Informational: request received, continuing to process the request	100 Trying, 180 Ringing
2xx	Successful: the action was successfully received, understood and accepted	200 OK
3xx	Redirection: further action needs to be taken in order to complete the request	302 Moved Temporarily
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

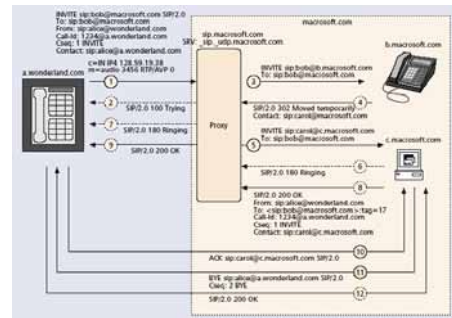
## Example: SIP Registration



## 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 = rtptime: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

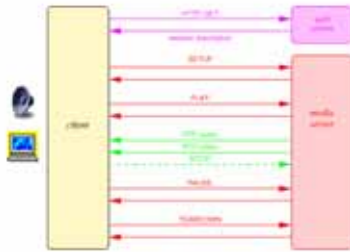
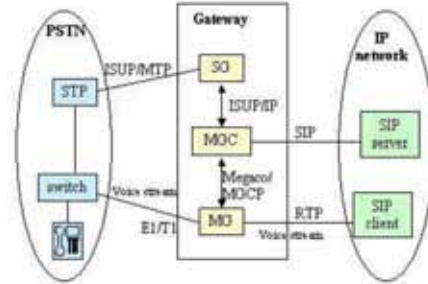


Figure 6: RTSP protocol session

## SIP Interworking with the SS7



- Introduction to SIP
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- Mobility management
- SIP and 3G Networks

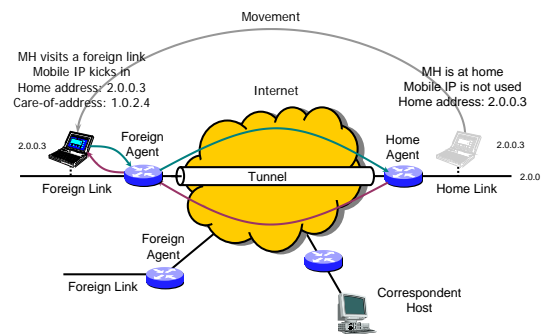
## Wireless Technologies Convergence



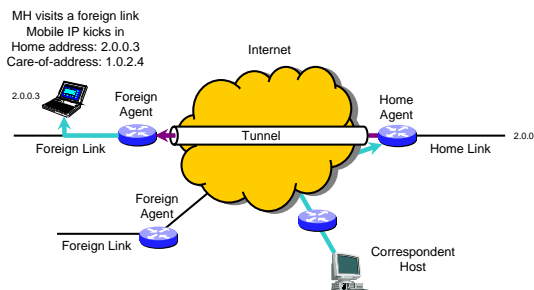
## Mobility Management

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

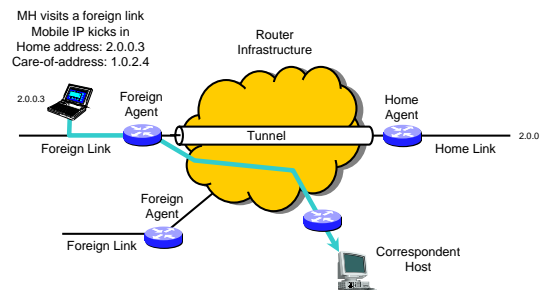
## Mobile IPv4: Registration Example



## Mobile IPv4: CH-to-MH Routing Example



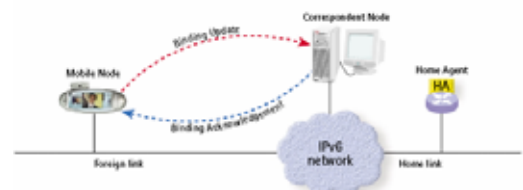
## Mobile IPv4: MH-to-CH Routing Example



## Mobile IPv4

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

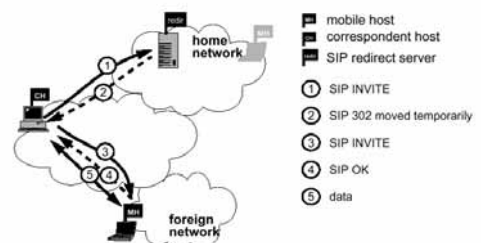
## Mobile IPv6: Binding Update



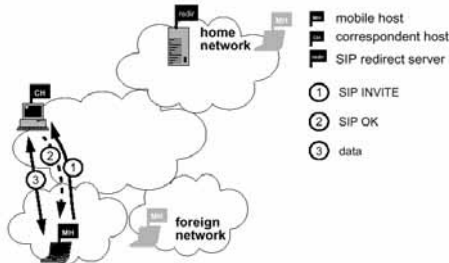
## Application Layer Mobility Using SIP

- Terminal Mobility
- Session Mobility

## Terminal Mobility

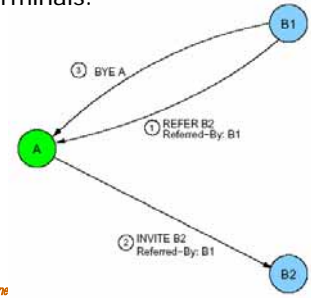


## Terminal Mobility

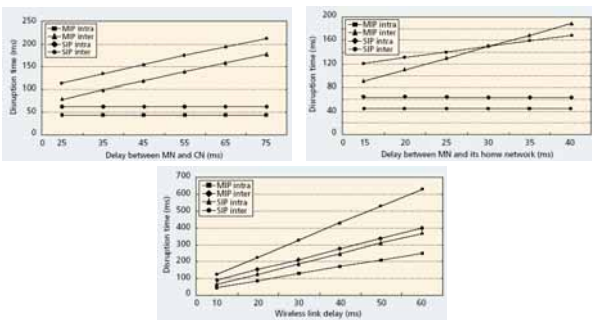


## Session Mobility

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

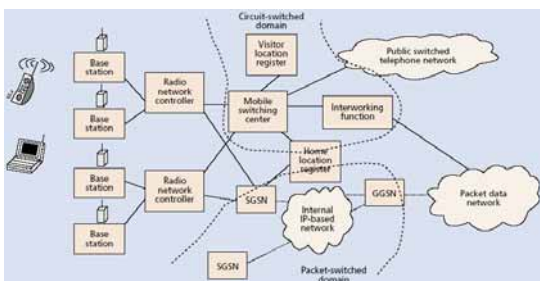


## Comparison



- Introduction to SIP
- SIP Architecture
- Internetworking
- Mobility management
- SIP and 3G Networks

## 3G UMTS



## 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 do 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.

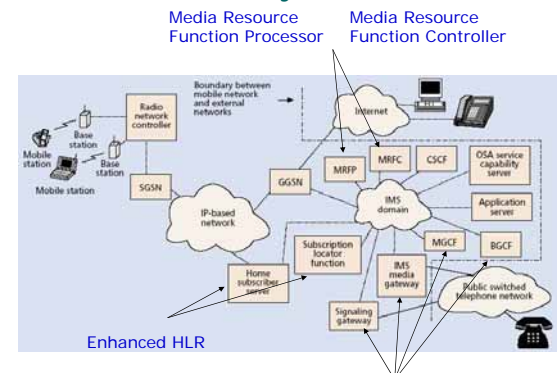
## IP Multimedia Subsystem (IMS)

- 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

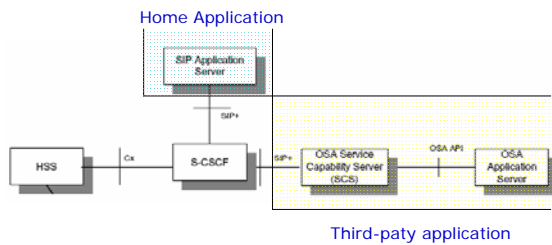
## Basic Add-in Features

- 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



## 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.

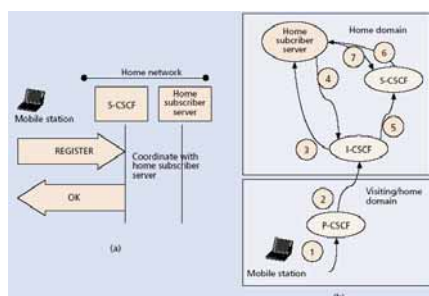
## Proxy-CSCF (P-CSCF)

- 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.

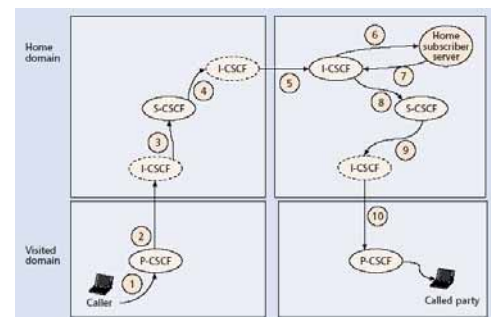
## Interrogating-CSCF (I-CSCF)

- 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





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## 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>

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## Thanks !!