

SIP: Session Initiation Protocol



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Outline

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

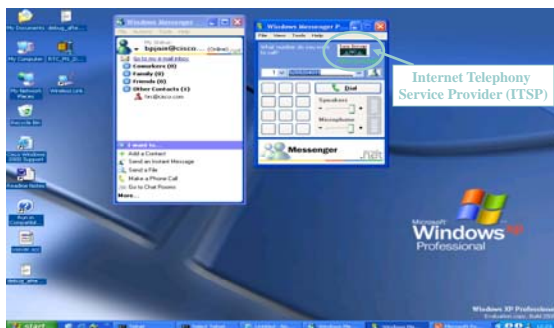
Session Initiation Protocol (SIP)



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 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.711/722, 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]

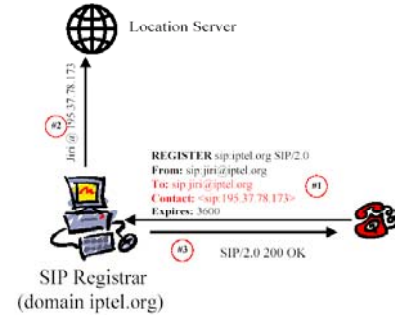
SIP message	Description
INVITE	Invites a user to a call
ACK	Used to facilitate reliable message exchange for INVITEs
OPTIONS	Solicits information about a server's capabilities
BYE	Terminates a connection between users or declines a call
CANCEL	Terminates a request, or search, for a user
REGISTER	Registers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

Request Line

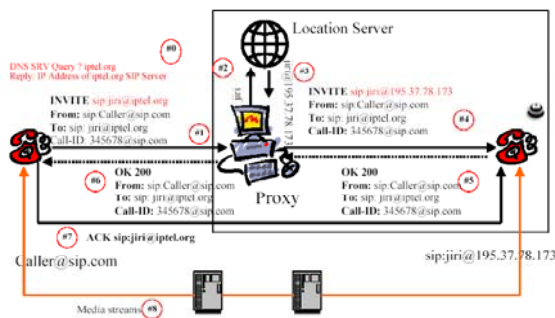
Class	Description	Example
1xx	Informational: request received, continuing to process the request	100 Trying, 180 Ringing
2xx	Successful: the action was successfully received, understood and accepted	200 OK
3xx	Redirection: further action needs to be taken in order to complete the request	302 Moved Temporarily
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 fulfill an apparently valid request	501 Not Implemented
6xx	Global Failure: the request cannot be fulfilled at any server	603 Decline

Status Line

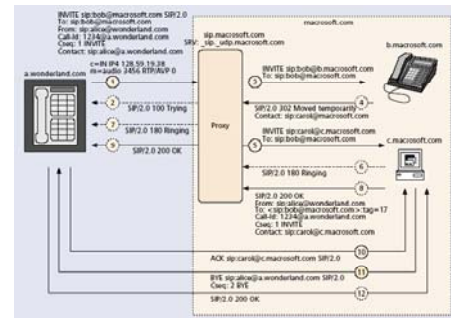
Example: SIP Registration



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 = 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

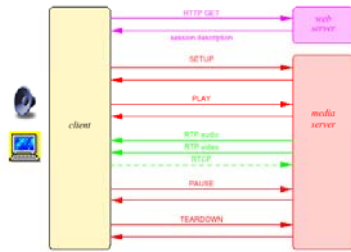
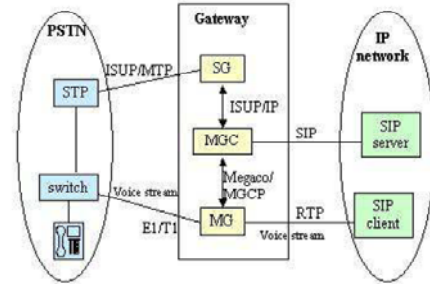


Figure 6: RTSP protocol session

SIP Interworking with the SS7



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

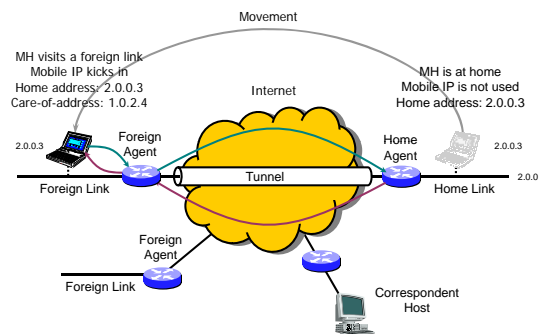
Wireless Technologies Convergence



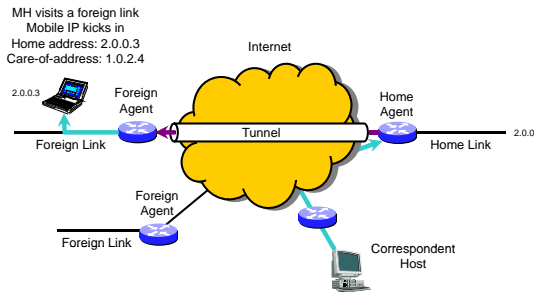
Mobility Management

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

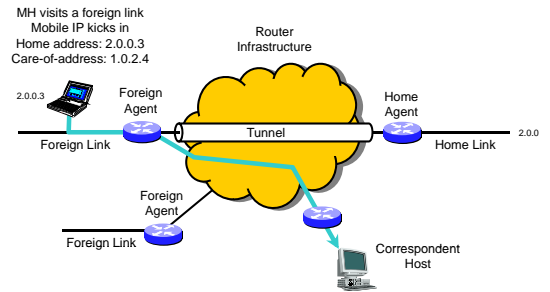
Mobile IPv4: Registration Example



Mobile IPv4: CH-to-MH Routing Example



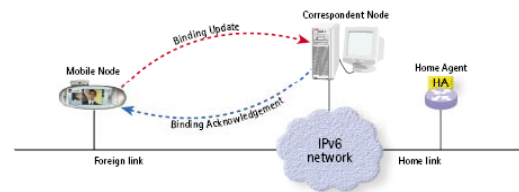
Mobile IPv4: MH-to-CH Routing Example



Mobile IPv4

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

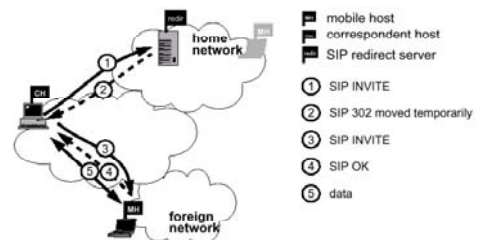
Mobile IPv6: Binding Update



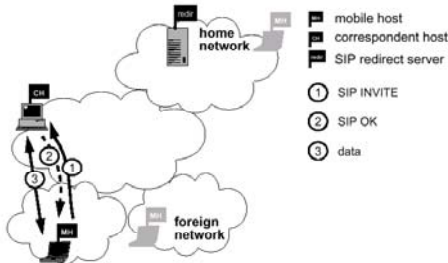
Application Layer Mobility Using SIP

- Terminal Mobility
- Session Mobility

Terminal Mobility

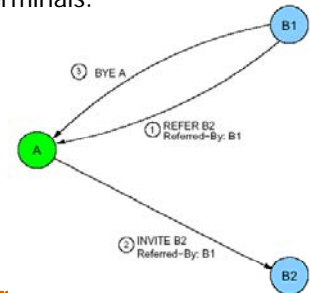


Terminal Mobility

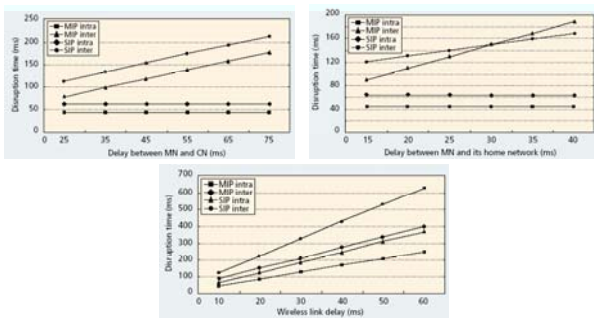


Session Mobility

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

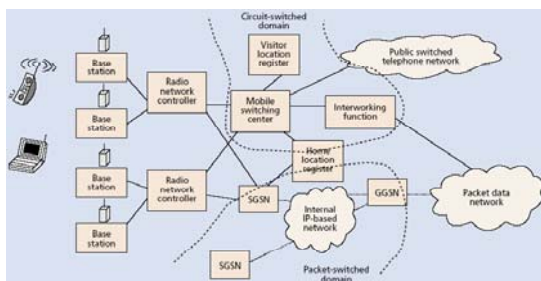


Comparison



- 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.

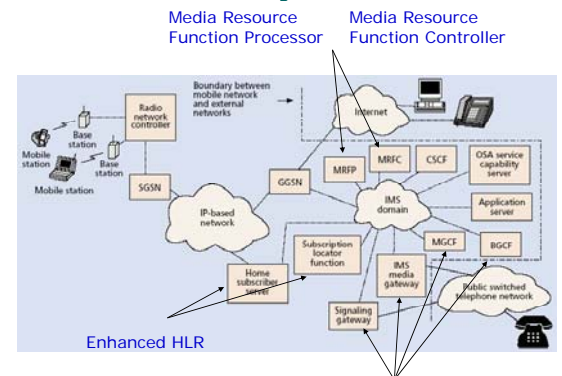
IP Multimedia Subsystem (IMS)

- 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

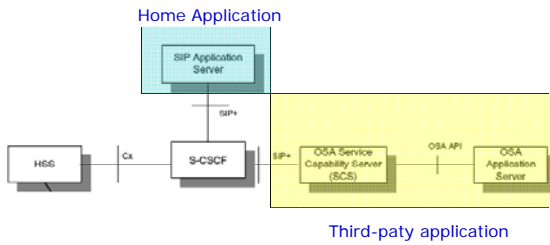
Basic Add-in Features

- 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.

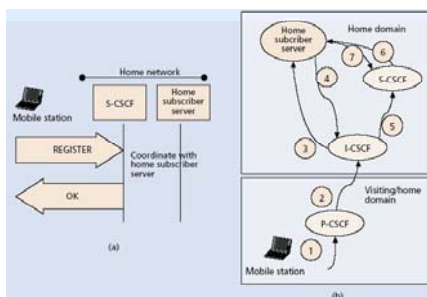
Proxy-CSCF (P-CSCF)

- 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.

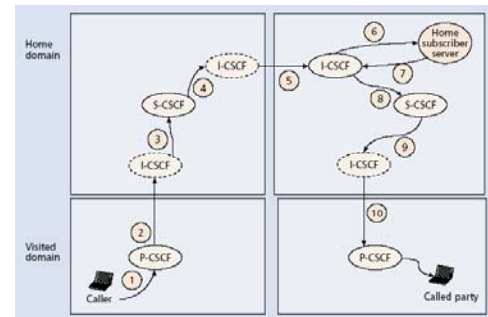
Interrogating-CSCF (I-CSCF)

- 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



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 !!