• Week #1
  • (January 4 - 8)
  • Current Generation Networks: From 2G to 2.5G
• Week #2
  • (January 11-15)
  • Value added Services in Current Generation Networks
• Week #3
  • (January 18-22)
  • Next Generation Network Vision
• Week #4
  • (January 25 – 39)
  • Web Services
  • Project specification available
• Week #5
  • (February 2 - 6)
  • Tools for Value Added Services Engineering (SDS and Web Logic)
• Week #6
  • (February 9 - 13) Quiz1
Chapter IV
SIP Session Signalling
And
SIP Specific Value Added Service Technologies
Part I
SIP Session Signaling
Outline

1. Introduction
2. Core SIP
3. Selected Extensions
Introduction: Signaling vs Media

Signaling:
- Session establishment
- Session tear down
- Changes to the session
- Supplementary services

Media:
- Actual communication data: encoded voice stream, video stream,...
Introduction: SIP

Signaling Protocols:
- SIP and H.323

Media transport protocol:
- RTP

Why SIP?

SIP: Prime signaling system because adopted by all key next generation networks:
- 3GPP
- 3GPP2
- PacketCable:
SIP: Introduction

A set of IETF specifications including:

- SIP core signalling:
  - RFC 2543, March 1999
  - RFC 3261, June 2002 (Obsoletes RFC 2543)

- SIP extensions (e.g. RFC 3265, June 2002 - Event notification)
  - May have nothing to do with signalling

- IMS related extensions.

- Used in conjunction with other IETF protocols
  - QOS related protocol (e.g. RSVP)
  - Media transportation related protocol (e.g. RTP - RFC 1889)
  - Others (e.g. SDP - RFC 2327)
Session Initiation Protocol (SIP) - Core

1. Introduction
2. Functional entities
3. Messages
4. SDP
5. Examples
SIP: Introduction

SIP core Signaling
- A signalling protocol for the establishment, modification and tear down of multimedia sessions
- Based on HTTP

A few key features
- Text based protocol
- Client/server protocol (request/response protocol)
SIP: The Request

Request messages
- Methods for setting up and changing sessions
  . INVITE
  . ACK
  . CANCEL
  . BYE

- Others
  . REGISTER (Registration of contact information)
  . OPTIONS (Querying servers about their capabilities)
SIP: The Response

Response message
- Provisional
- Final

Examples of status code
1xx: Provisional
2xx: Success
6xx: Global failure
SIP: A basic peer to peer call scenario

CALLER

INVITE

100 TRYING

180 RINGING

200 OK

ACK

MEDIA SESSION

BYE

200 OK

CALLEE
SIP: The functional entities

User agents
- End points, can act as both user agent client and as user agent server
  - User Agent Client: Create new SIP requests
  - User Agent Server: Generate responses to SIP requests

Proxy servers
- Application level routers

Redirect servers
- Redirect clients to alternate servers

Registrars
- Keep tracks of users
SIP: The functional entities

State-full proxy

- Keep track of all transactions between the initiation and the end of a transaction

- Transactions:
  - Requests sent by a client along with all the responses sent back by the server to the client

Stateless proxy

- Fire and forget
SIP: A call scenario

CALLER | PROXY A | PROXY B | CALLEE
---|---|---|---
INVITE (1) | INVITE (2) | INVITE (4) | CALLEE
100 TRYING (3) | 100 TRYING (5) | 180 RINGING (6) |
180 RINGING (8) | 180 RINGING (7) | 200 OK (9) |
200 OK (11) | 200 OK (10) | ACK (12) |
| MEDIA SESSION | | |
| | | BYE (13) |
| | | 200 OK (14) |

Roch H. Glitho
SIP: The messages

Generic structure
- Start-line
- Header field(s)
- Optional message body

Request message
- Request line as start line
  . Method name
  . Request URI
  . Protocol version

Response message
- Status line as start line
  . Protocol version
  . Status code
  . Reason phrase (Textual description of the code)
SIP: Examples of messages from the RFC

An example of an INVITE

```
INVITE sip:bob@biloxi.com SIP/2.0
 Via: SIP/2.0/UDP
 pc33.atlanta.com;branch=z9hG4bK776asdhds
 Max-Forwards: 70
 To: Bob <sip:bob@biloxi.com>
 From: Alice <sip:alice@atlanta.com>;tag=1928301774
 Call-ID: a84b4c76e66710@pc33.atlanta.com
 CSeq: 314159 INVITE
 Contact: <sip:alice@pc33.atlanta.com>
 Content-Type: application/sdp
 Content-Length: 142
```
SIP: Examples of messages from the RFC

An example of RESPONSE to the OPTIONS request
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKhjhs8ass877 ;received=192.0.2.4
To: <sip:carol@chicago.com>;tag=93810874
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 63104 OPTIONS
Contact: <sip:carol@chicago.com>
Contact: <mailto:carol@chicago.com>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE
Accept: application/sdp
Accept-Encoding: gzip
Accept-Language: en
Supported: foo
Content-Type: application/sdp
SDP

Session Description Protocol
- Convey the information necessary to allow a party to join a multimedia session
  - Session related information
  - Media related information
- Text based protocol

- No specified transport
  - Messages are embedded in the messages of the protocol used for the session
    - Session Announcement Protocol (SAP)
    - Session Initiation Protocol (SIP)
SDP

Session Description Protocol

Use with SIP

- Negotiation follows offer / response model
- Message put in the body of pertinent SIP messages
  INVITE Request / response
  OPTIONS Request / response
SDP

Session Description Protocol
- `<Type> = <Value>`
- Some examples
  Session related
    v= (protocol version)
    s= (Session name)
  Media related
    m= (media name and transport address)
    b= (bandwidth information)
SDP: Examples of messages from the RFC

Session Description Protocol
An example from the RFC ...

v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
s=SDP Seminar
i=A Seminar on the session description protocol
u=http://www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps
e=mjh@isi.edu (Mark Handley)
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
a=recvonly
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
m=application 32416 udp wb
a=orient:portrait
SIP – Selected Extensions

1. Event framework
2. INFO method
Event Notification

Motivation

- Necessity for a node to be asynchronously notified of happening(s) in other nodes
  - Busy / not busy (SIP phones)
    - A client A can call again a client B when notified that B is now not busy
  - On-line / Off-line
    - Buddy list
Event Notification

Conceptual framework

Requestor

Subscribe (specific event(s))

Notify (specific event)

Notify (specific event)

Notify (specific event)

Un-subscribe (specific event(s))

Provider
Event Notification

The SIP Event Notification Framework

- Terminology
  - Event package:
    - Events a node can report
    - Not part of the framework – Part of other RFCs
  - Subscriber
  - Notifier

- New Messages
  - Subscribe
    - Need to be refreshed
    - Used as well for un-subscribing (expiry value put to zero)
  - Notify
Event Notification

The SIP Event Notification Framework
- More on the methods
  - New headers
    - Event
    - Allow-Events
    - Subscription state
Event Notification

An example of use: REFER Method

- Recipient should contact a third party using the URI provided in the CONTACT field
  - Call transfer
  - Third party call control
- Handled as Subscribe / notify
  - REFER request is considered an implicit subscription to REFER event
    - Refer-TO: URI to be contacted
    - Expiry determined by recipient and communicated to sender in the first NOTIFY
    - Recipient needs to inform sender of the success / failure in contacting the third party
Event Notification

Another example of use: Presence
- Dissemination/consumption of presence information (e.g. on/off, willingness to communicate, device capabilities, preferences)
  - Numerous applications
    - Multiparty sessions initiated when a quorum is on-line
    - News adapted to device capabilities
- Several standards including SIMPLE (SIP based)
  - Handled as Subscribe / notify in SIMPLE
    - Watchers / presentities
      - Explicit subscriptions
      - Explicit notifications
INFO Method

Allow the exchange of non signalling related information during a SIP dialog
- Semantic defined at application level
- Mid-call signalling information
  - DTMF digits with SIP phones
- Info carried as
  - Headers and/or
  - Message body
References

Core SIP
- SIP core signalling:
- H. Schulzrinne, an J. Rosenberg, SIP: Internet Centric Signaling, IEEE Communications Magazine, October 2000
- RFC 3261, June 2002 (Obsoletes RFC 2543)
- RFC 2327 (SDP)

SIP extensions
No overview paper
- RFC 3265, 3515 (Event framework)
- RFC 2976 (INFO Method)
Part II
SIP Specific Value-Added Service Technologies
By Dr Hechmi Khilfi
SIP Specific Value Added Service Technologies

1. Introduction: SIP specific architectures vs protocol neutral architectures
2. SIP CGI
3. SIP servlet API
Introduction: SIP specific architectures

- Servers built using SIP specific architectures act as redirect servers, proxy servers, originating user agents, terminating user agents, or back-to-back user agents.
- They have SIP signaling capabilities and are directly involved in the call’s signaling flow.
- Implementation techniques: SIP CGI, SIP Servlet
Introduction: Protocol neutral architectures

- Servers built using protocol neutral architectures can provide the same services as the SIP application server, but are:
  - signaling independent (i.e. could be used with any signaling protocol).
  - Are not directly involved in the SIP calls’ signaling flow.
- Examples of APIs: TAPI, TSAPI, JTAPI, Parlay and Web services/Parlay X
  - Focus of this lecture: SIP specific value added services technologies (i.e. SIP application servers)
  - Web services / Parlay-X will be discussed in another lecture
SIP CGI

Key features
- Inspired by HTTP CGI
- The server passes the message body to the script through its standard input
- Services are written as CGI scripts
SIP CGI: shortcomings

- Difficult to program
- Require a deep understanding of SIP protocol
SIP Servlet: Introduction

Key features

– Signalling protocol specific (i.e. applicable to SIP only)
– Prime target: trusted parties
  • Service providers
  • Third party developers
– Very few constraints on what can be done
– Reliance on HTTP servlet API
  • HTTP servlet API is widely used in the Internet world
    – A tool which relies on it should attract many users including Web
      masters.
    – A wide range of developers should favour the development of cool
      and brand new services


HTTP servlet API ...

Creation of dynamic Web content

- Servlet
  - Java component
  - Generate content on the fly, just like HTTP CGI
    - interface between HTTP request and data bases
    - Forms
    - Dynamic information (e.g. date, number of visitors)
HTTP servlet API ...

Servlet container (also know as servlet engine)

- Servlet container (or servlet engine)
  - Contains the servlets
  - Manage the servlets through their life cycle
    - Creation
    - Initialisation
    - Destruction
    - Receives and decodes of HTTP requests
    - Encodes and sends of HTTP responses
HTTP servlet API ...

Pros

Address most HTTP CGI shortcomings
- Performance
  - Can keep data base connections open
- Scalability
  - Servlet containers can be accessed remotely

Cons

• Language dependence
SIP servlet API...

Adjustments made to HTTP servlet:

- Initiate requests
  - Needed for some services
    - wake up call
- Receive both requests and responses
  - Needed for some services
    - Terminating services (e.g. call forward on busy)
- Possibility to generate multiple responses
  - Intermediary responses, then final response
- Proxying requests, possibly to multiple destinations
  - Needed for applications such as intelligent routing
SIP Servlet container ... 

A container collocated with a proxy server
SIP servlet Request interface ...

SIP specific Request handling methods (Based on both core SIP and SIP extensions):

- doInvite
- doAck
- doOptions
- doBye
- doCancel
- doRegister
- doSubscribe
- doNotify
- doMessage
- doInfo
SIP servlet Response interface ...

SIP specific Response handling methods (Based on both core SIP and SIP extensions):

- doProvisionalResponse
- doSuccessResponse
- doRedirectResponse
- doErrorResponse
An example of service:

**Algorithm for call forward**

- Get the destination from the SIP request
  - Done by retrieving the To_Field by using the GetHeaders
- Obtain the forwarding address from a data base
- Forward the call
  - Done by setting the Request_URI (and not the To_field) using the setHeader
Another example:

Algorithm for a centralized dial-out conference

Assumptions
– INVITE is used
– URIs of participants are put in the INVITE body

Algorithm used in servlet:
• Use GetContent to get the participant’s URIs from INVITE Request
• Use doINVITE to generate and send an INVITE to each participant.
public class RegistrarServlet extends SipServlet{

    protected void doRegister(SipServletRequest request) throws ServletException, IOException {
        SipServletResponse response = request.createResponse(200);
        response.send(); logger.log(Level.FINE, "Sent 200 response.");
    } catch(Exception e) {

        response.setStatus(500); response.send();
    }
}


Pros and cons

Pros
- Possibility of creating a wide range of services due to the full access to all the fields from the SIP Request
- More performance and more scalability
- Possibility to create services that combine both HTTP and SIP

Cons:
- SIP Servlet is not exactly the same thing as HTTP Servlet
- Language dependence
References
