Telecommunication Services Engineering (TSE) Lab

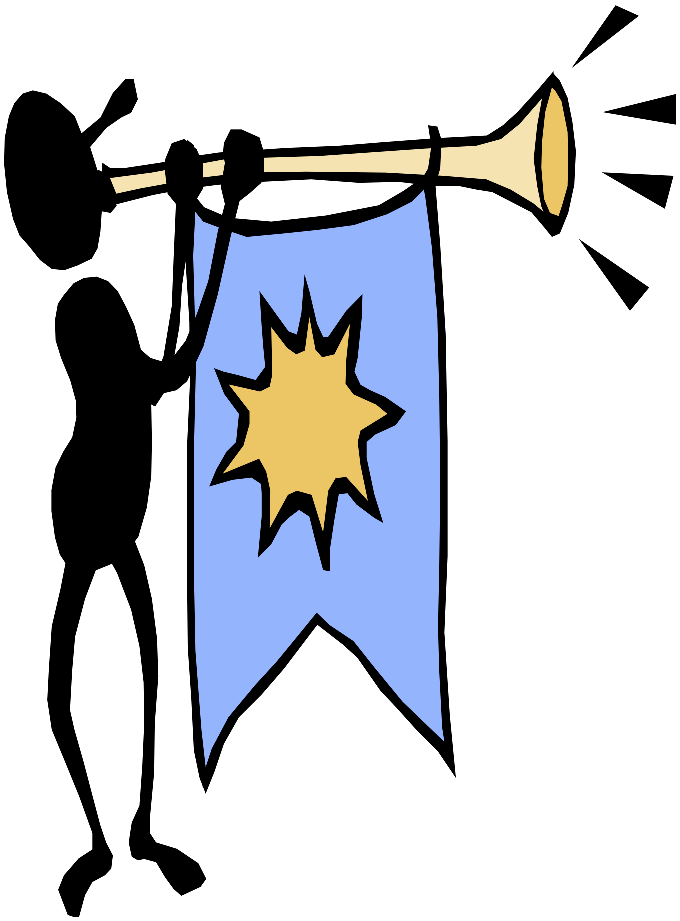
Chapter V –

SIP Technology For Value Added Services (VAS) in NGNs

http://users.encs.concordia.ca/~glitho/

Concordia University
Engineering and Computer Science
Concordia Institute for Information Systems Engineering

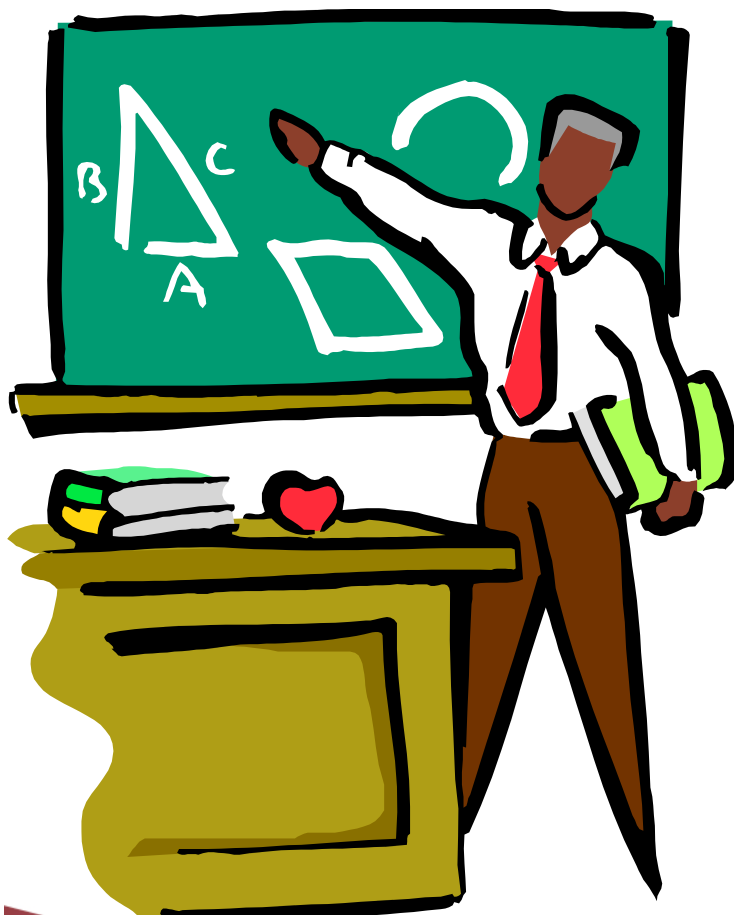
Outline



1. SIP
2. SIP servlets
3. Examples of services that may be implemented with SIP technology

Telecommunication Services Engineering (TSE) Lab

SIP – Core



1. Introduction
2. Functional entities
3. Call scenario
4. SDP

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)

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)

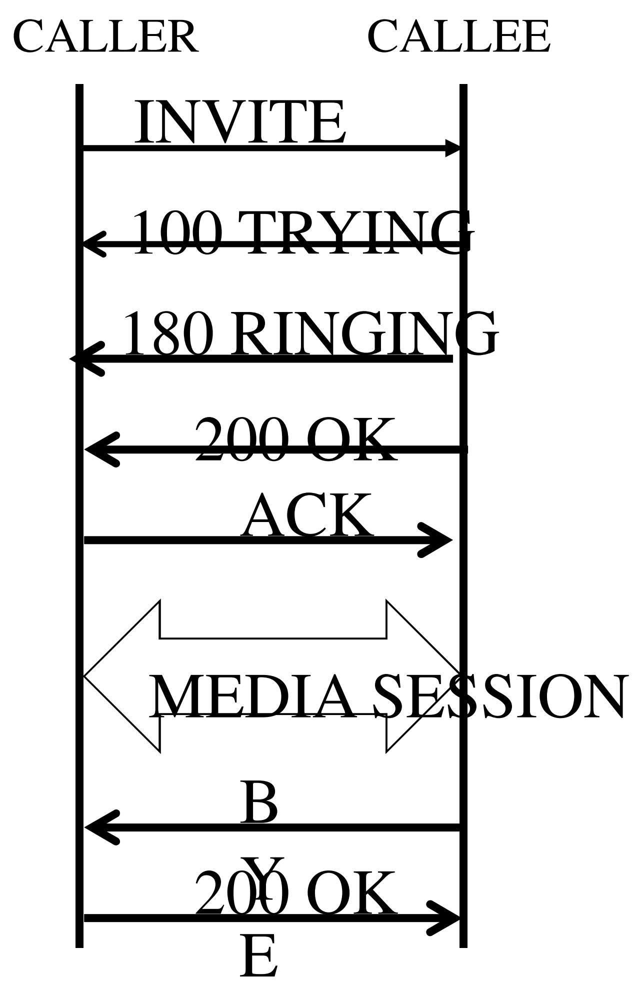
SIP: The Response

Response message

- Provisional
- Final

Examples of status code
1xx: Provisional
2xx: Success
6xx: Global failure

SIP: A basic peer to peer call scenario



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

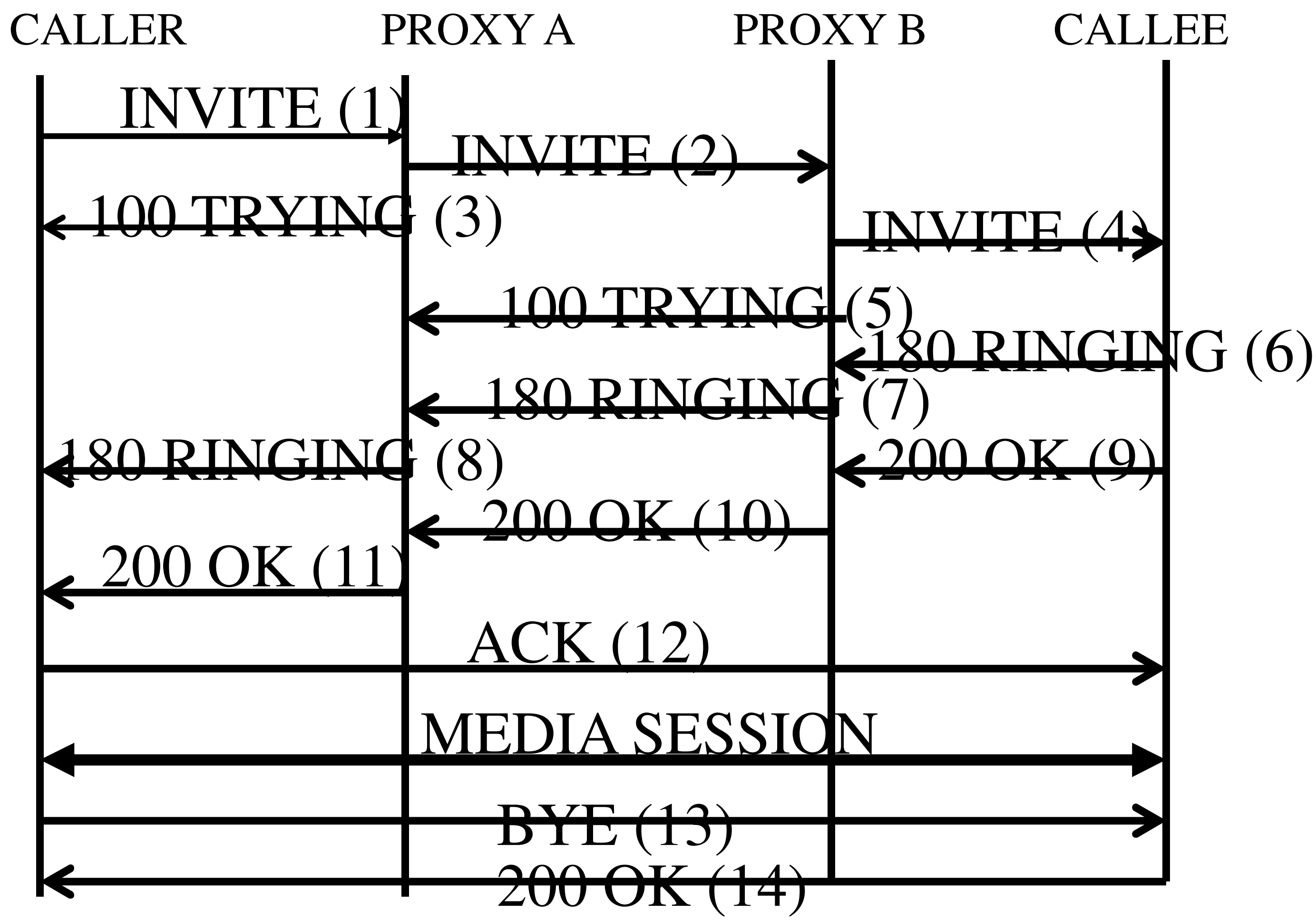
Redirect servers

- Redirect clients to alternate servers

Registrars

- Keep tracks of users

SIP: A call scenario



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 …

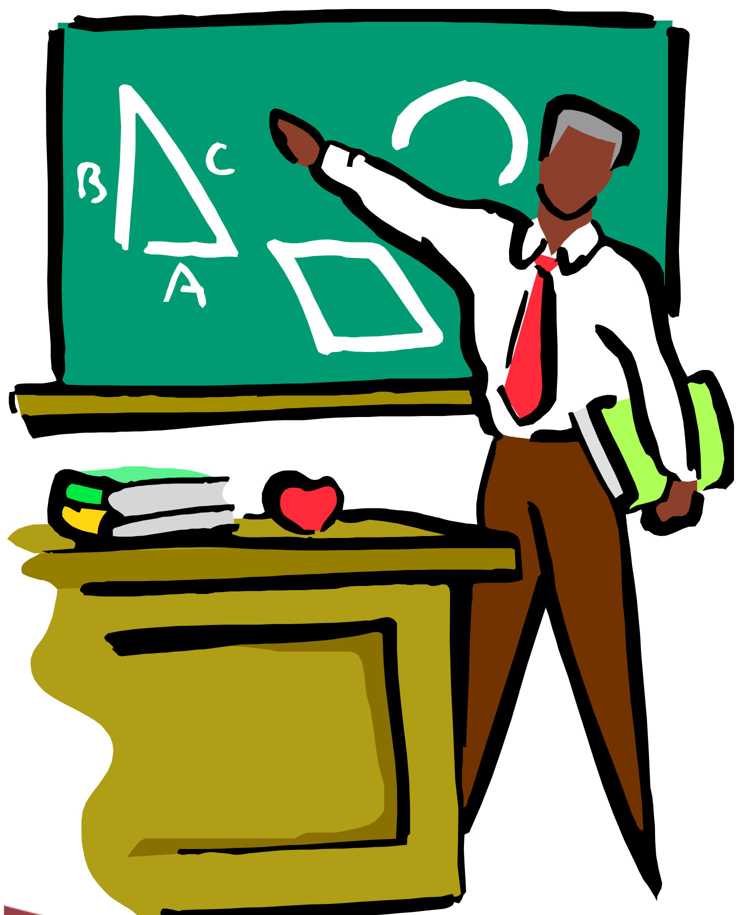
Session Description Protocol

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
```

Telecommunication Services Engineering (TSE) Lab

SIP – Selected Extensions



1. Event framework
2. INFO method

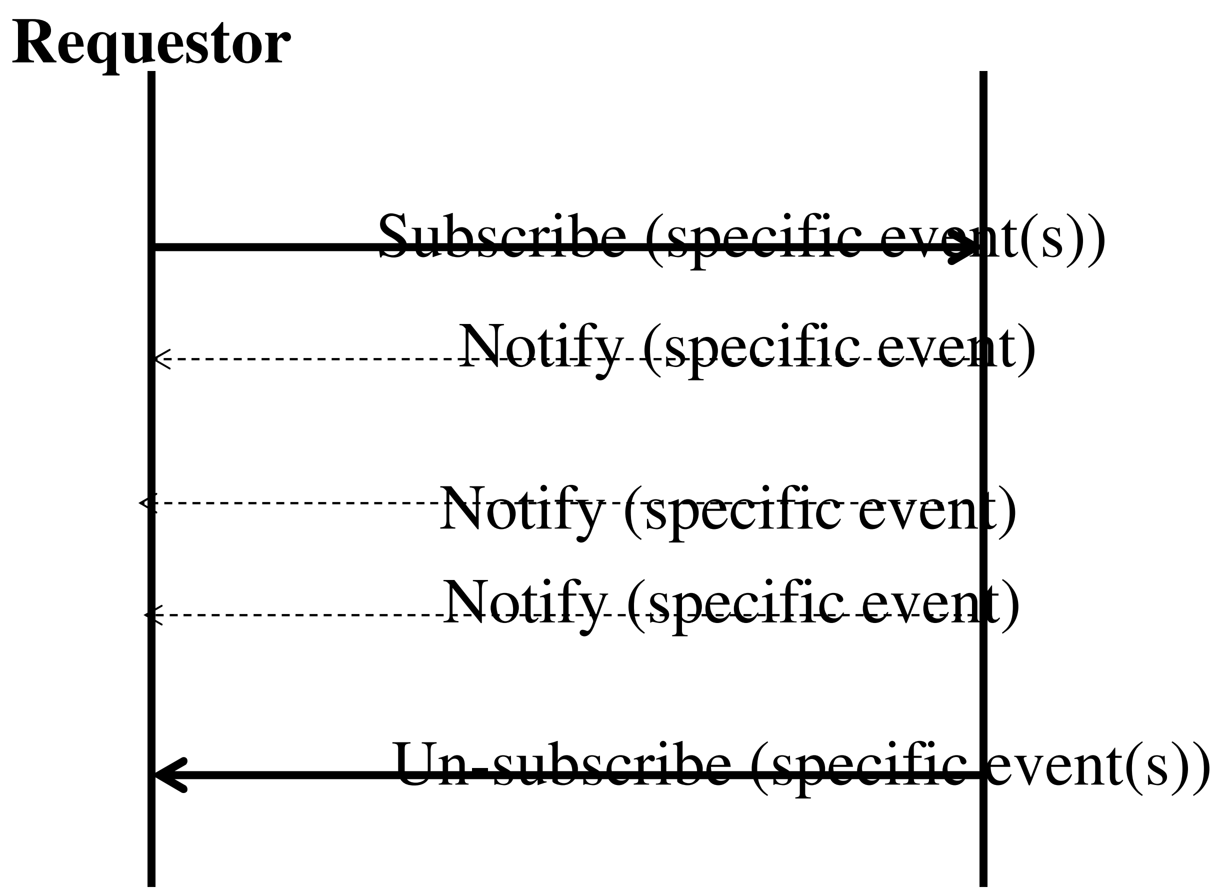
Event Notification

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



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

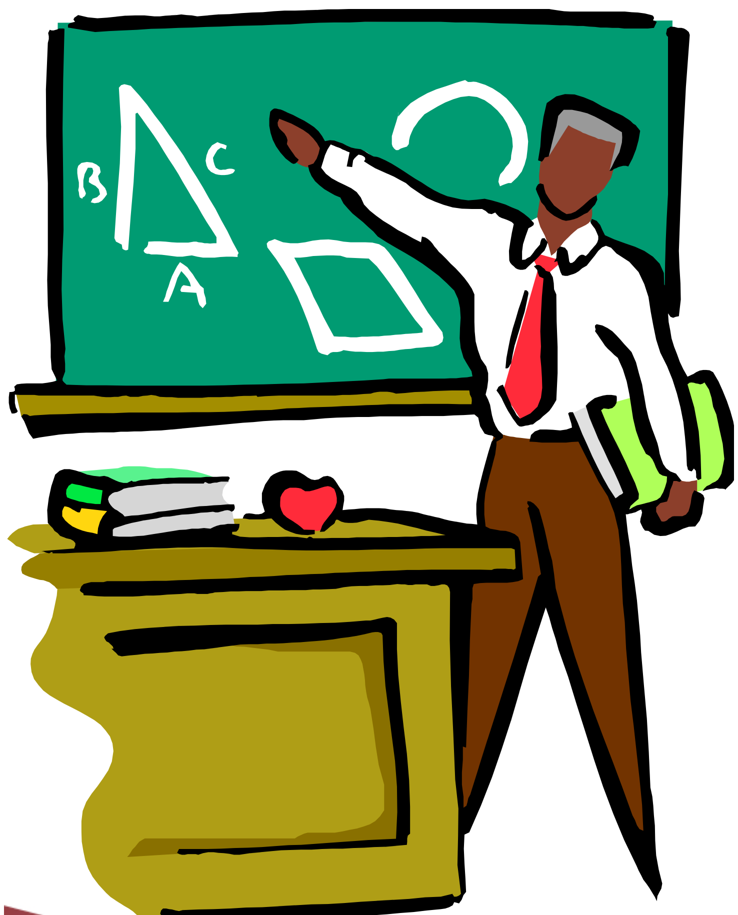
INFO Method

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

Telecommunication Services Engineering (TSE) Lab

SIP Specific Value Added Service Technologies



1. Introduction: SIP specific architectures vs protocol neutral architectures

2. SIP CGI

3. SIP servlet API

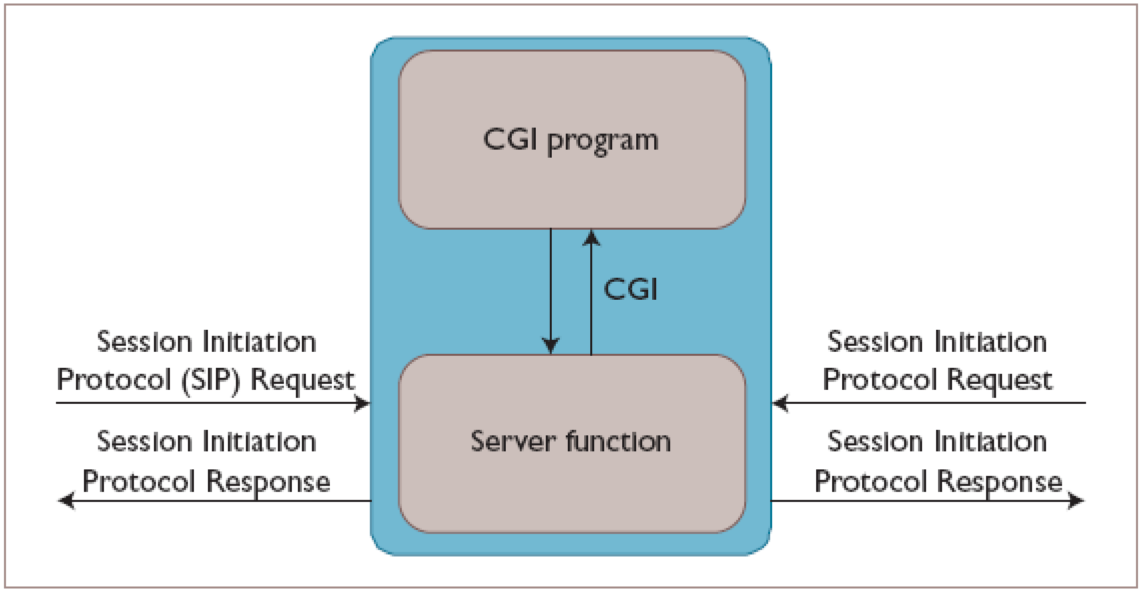
Introduction: SIP specific architectures

- 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: SOAP Based Web services/Parlay X, RESTFul Web services
  - 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...

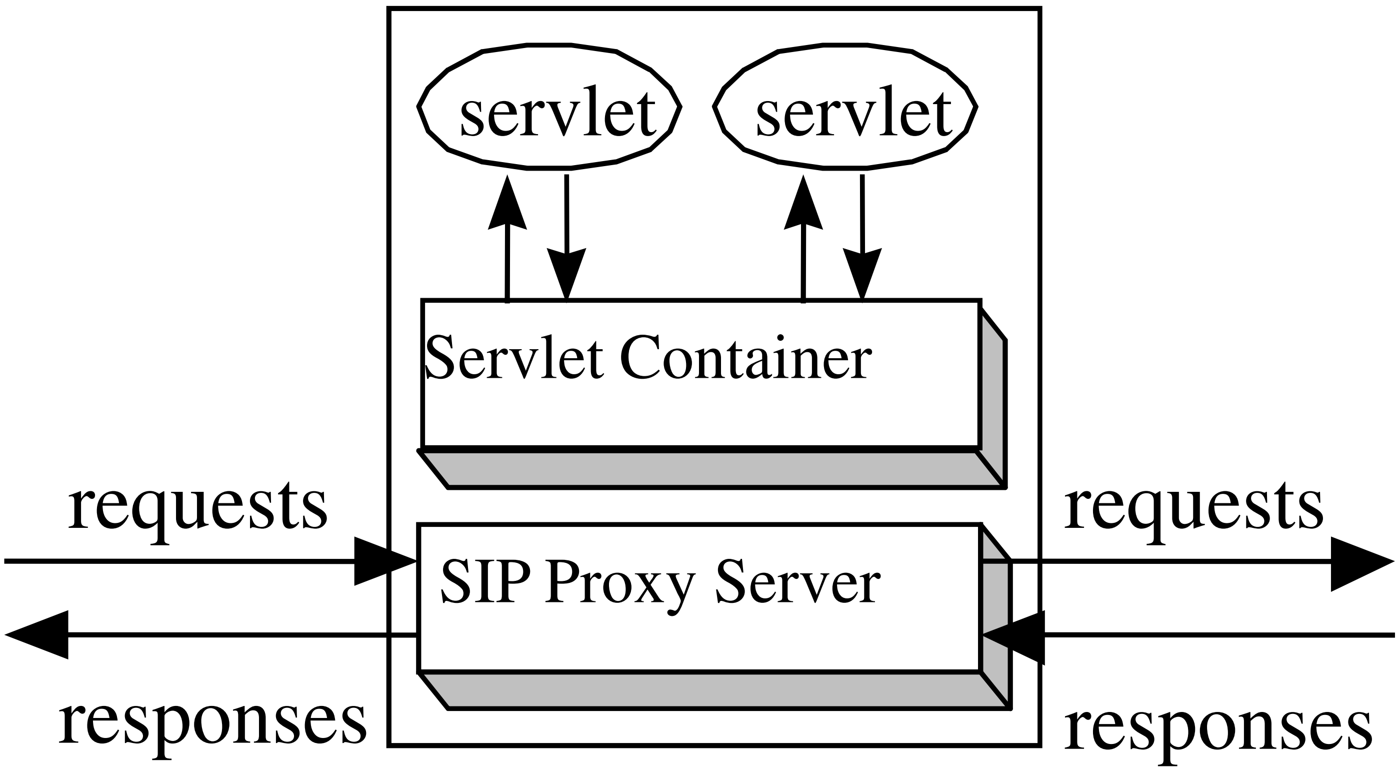
Adjustments made to HTTP servlet:

- 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 ...

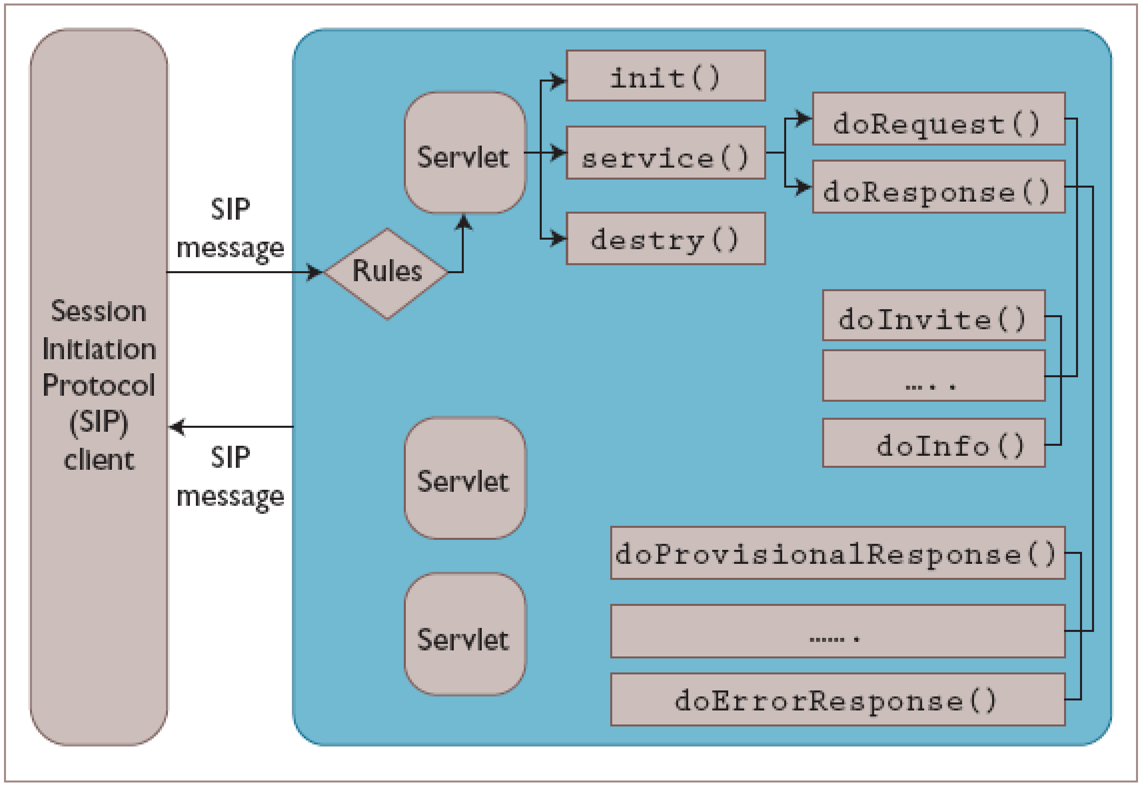
A container collocated with a proxy server



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

Agorithm 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.

Example

```
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

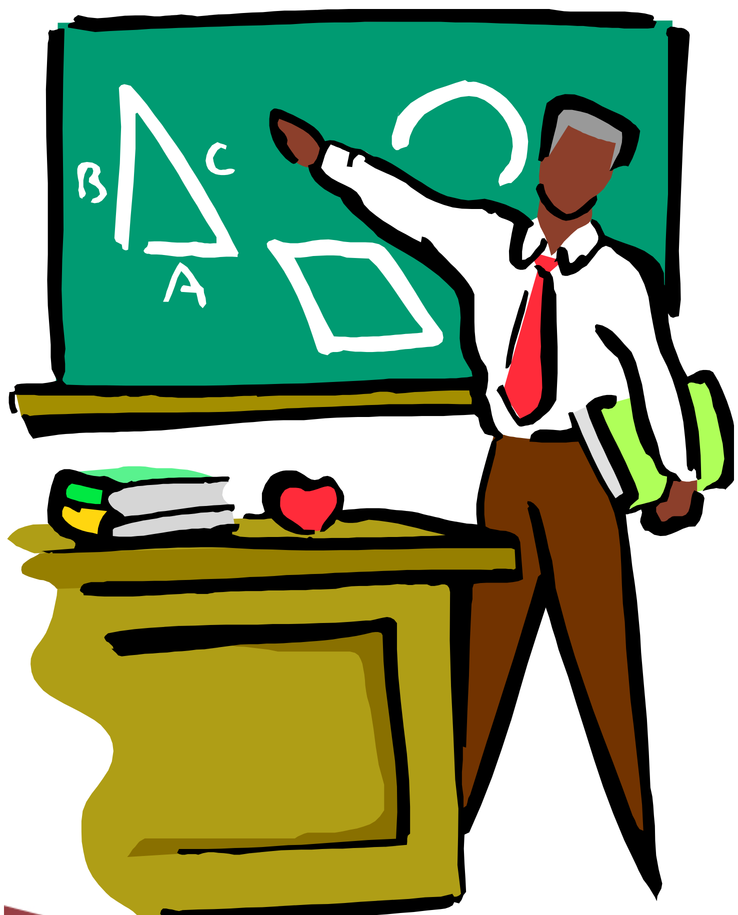
- Possibility of creating a wide range of services due to the full access to all the fields from the SIP Request
- More performance and more scalability
- Possibility to create services that combine both HTTP and SIP

Cons:

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

.

Examples of Services that may be implemented with SIP



Examples of services

- Call transfer
- Call diversion
- Call hold
- Call park and pick up
- Call waiting
- Message waiting indication
- Name identification
- Call completion
- Call offer
- Call intrusion

Examples of services

Call transfer

Allow a user A in communication with user B to establish a new call between user B and user C

- First case: User A has a call established with user C before the transfer
- Second case: User A does not establish any call with user C before the transfer

Call diversion

Divert the call (before answering it) if some conditions are met

- Unconditional
- Busy
- No reply

Examples of services

Call hold

Allow a user A to put user B on “hold” after the call has been established

- User B can hear music / advertisement in the meantime

Also allow user A to retrieve a call previously put on hold

Call park and pick up

Generalization of call hold / retrieve

- Parking places (I.e identifier for each parked call)
- Retrieval using identifiers

Examples of services

Call waiting

Allow a busy user to be notified of an incoming call and to decide how to proceed (Classical example; Internet call waiting)

- Accept (I.e give up on previous call)
- Reject
- Divert

Message waiting indication

Self explanatory

- User can call a message center

Examples services

Name indication

Self explanatory …

Call completion

Camp on ….

- Allow caller to establish a call with a busy callee as soon as callee is free and without having to re-dial callee's number.

Examples of services

Call offer …

Strong form of call completion

Allow caller to offer a call to a busy callee and wait till busy callee accepts the call …

Call intrusion

Allow user A to establish a call with a busy user B by breaking into the call between B and C

- Result: 3 party call

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)**