A SIP-based Architecture for Internet Telephony

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SIP

- SIP = modular, programmable, web-centric protocol for creating multimedia sessions
- Internet standard
- wide industry support: 50 implementations at 4th SIP bake-off
- will be used for 3rd generation wireless network
- customer network trials likely later this year
SIP architecture

- goal: construct a complete SIP-based Internet telephony suite
  - e*phone as PBX replacement
  - SIP server for creating services
  - multimedia client for workstations and PCs
  - library for building new applications
  - SIP multiparty conferencing application
  - telephone gateway
  - standards-based unified messaging
  - location-based SIP services
SIP phone: e*phone

- first SIP-based Ethernet phone
- same device also works as MP3 radio
- sensor interfaces to the world: IR, temperature, alarms, medical measurements, ...
SIP server (sipd)

- core component for routing SIP calls
- translates names and numbers
- filters and routes calls, e.g., based on caller, time of day, urgency, ...
- programmable services via web-like scripting
- 90 academic and commercial licenses
SIP client (sipc)

- cross-platform client for creating multimedia sessions
- any media type can be added via external programs, including audio, video, chat, whiteboard, ...
- feature programming and calendar interfaces in progress
SIP-H.323 gateway

- allow SIP devices to communicate with H.323 devices, such as Microsoft NetMeeting
- single server sufficient for whole organization
SIP conferencing bridge

- set up conference via web page
- allow “dial-in” conferences
- mixes audio (and video) streams to all participants
- can be used locally or offered as a service
SIP unified messaging

- calls are redirected to SIP messaging server
- server instructs media server to record – any media (audio, video, chat, …)
- can be retrieved from web page or via email link
- use any RTSP player, e.g., QuickTime, to play messages from anywhere
### SIP unified messaging – web interface

**VoiceMail**

<table>
<thead>
<tr>
<th>Inbox</th>
<th>Folders</th>
<th>Options</th>
<th>Help</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
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</tr>
</tbody>
</table>

#### Messages

<table>
<thead>
<tr>
<th>Date</th>
<th>From</th>
<th>Subject</th>
<th>Size</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mon 01:32 PM</td>
<td><a href="mailto:sip.wxt@sirr.cs.columbia.edu">sip.wxt@sirr.cs.columbia.edu</a> email</td>
<td>[normal] Hi</td>
<td>14 s  (119 KB)</td>
</tr>
<tr>
<td>Mon 01:06 PM</td>
<td>sip.12125551112@128.59.19.193 email</td>
<td>Message: 9164.au</td>
<td>0 s   (0 KB)</td>
</tr>
<tr>
<td>Mar 29</td>
<td><a href="mailto:sip.user1@cs.columbia.edu">sip.user1@cs.columbia.edu</a> email</td>
<td>test call</td>
<td>6 s   (53 KB)</td>
</tr>
<tr>
<td>Feb 28</td>
<td><a href="mailto:sip.user1@cs.columbia.edu">sip.user1@cs.columbia.edu</a> email</td>
<td>Message: 3513.au</td>
<td>4 s   (32 KB)</td>
</tr>
</tbody>
</table>

**Buttons:**
- Delete Selected
- Move to Folder
- Refresh
- Change User
- Permanent Signout

**Forward** selected mails to these email(s): I