The Columbia University SIP Suite: CINEMA

Henning Schulzrinne
Internet Real-Time Lab
Dept. of Computer Science
Columbia University
New York, New York
http://www.cs.columbia.edu/IRT
schulzrinne@cs.columbia.edu

January 2001
Overview

- architecture
- SIP library: C/C++ and Java
- SIP clients
- SIP proxy/redirect/application server
- SIP applications: unified messaging, conferencing
- related work: QoS, charging, mobility, 911 services
Goals

- commercial-grade, but simple implementations
- standards-compliant:
  - SIP (RFC 2543 and some extensions)
  - Call Processing Language (CPL)
  - sip-cgi
- cross-platform:
  - Unix: Solaris, Linux, FreeBSD, True64, ...
  - Windows 98, NT and 2000
- use open-source components where feasible:
  - mySQL for SQL database (user configuration)
  - OpenLDAP
common code base for everything except sipc and e*phone
Columbia University Computer Science test bed

- Columbia CS runs its own PBX (15-year old Nortel Meridian)
- allow both intra-department and external in/out calls
Campus network

Cisco-based (6509) gigabit Ethernet:

At least 4 switched jacks in each office.
Columbia University CS conference room

- Video cameras
- PC
- Automatic mixer
- Laptop (on table)
- Speakerphone
- VCR
- CD
- DAT
- Projector
- Amplifier

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e*phone

DSP-based, single-processor Ethernet phone; being commercialized
SIP user agent sipc
Device control

SIP user agent

SUBSCRIBE door@alice.home.net
NOTIFY alice@work.com
DO light@alice.home.net
INVITE camera@alice.home.net

SIP proxy (RGW)

January 2001
sipd – SIP proxy, registrar and redirect server

- core call routing and feature component
- performance-optimized parser and implementation
- Apache-like configuration file and request logging
- basic, digest and PGP authentication for calls and registration
- user information stored in SQL database
- name resolution via access to LDAP directory
- supports SIP cgi and CPL (soon) for implementing features
- canonicalization of names: John.Doe, J_Doe, J.Doe, doe, ... → jd123
- translation of tel URLs and dial-plans
Email send to henning@cs.columbia.edu:

Subject: Your CINEMA registration
Date: Tue, 24 Oct 2000 21:48:09 -0400 (EDT)
From: <CGI.script.-.do.not.reply@cs.columbia.edu>
To: henning@cs.columbia.edu

Your new CINEMA password for cs.columbia.edu is "deduct.transversal.desert".
The realm is "Password for cs.columbia.edu".
sipd single sig-on

Please provide CINEMA user information for cs.columbia.edu

User name (user@host, must be a valid email address):

henning@cs.columbia.edu

Realm (used for prompting):

Your cs.columbia.edu password

Add user
sipd user configuration

SIP User List

<table>
<thead>
<tr>
<th>User name</th>
<th>Role</th>
<th>Groups</th>
<th>Authentication</th>
<th>Algorithm</th>
<th>SIP methods</th>
<th>Allows</th>
<th>Contacts</th>
<th>Delete?</th>
<th>Users that can register for this user</th>
<th>Last modified</th>
</tr>
</thead>
<tbody>
<tr>
<td><a href="mailto:default@cs.columbia.edu">default@cs.columbia.edu</a></td>
<td>Password for cs.columbia.edu</td>
<td>cs, minimal</td>
<td>request</td>
<td>MD5</td>
<td>REGISTER INVITE</td>
<td></td>
<td></td>
<td></td>
<td><a href="mailto:33310@cs.columbia.edu">33310@cs.columbia.edu</a>, <a href="mailto:hegb@cs.columbia.edu">hegb@cs.columbia.edu</a>, <a href="mailto:lng@cs.columbia.edu">lng@cs.columbia.edu</a></td>
<td>12 Oct 2000</td>
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<td>12 Oct 2000</td>
</tr>
<tr>
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<td>12 Oct 2000</td>
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<tr>
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</table>
SIP unified messaging

- receives SIP calls: forwarding, forking proxy
- uses RTSP server to play back announcement
- record audio (and, later, video) from caller
- playback through any RTSP-capable client, such as Real, QuickTime, ...
SIP conferencing

- multipoint control unit for audio and video conferences
- mixes audio, replicates video packets
- “dial in”: just dial faculty-meeting@cs.columbia.edu
SIP-H.323 translation

- allow SIP devices to *transparently* call H.323 systems (e.g., NetMeeting)
- allow H.323 to call SIP devices
- serves as H.323 GK to register H.323 participants with SIP registrar
Some future plans

Java SIP library: not just parser, but complete SIP & SDP implementation for UAs and proxies

sipc: instant messaging & presence

Conferencing: video distribution, floor control, conference timing

Unified messaging: access from POTS, VPIM

sipd: CPL, performance evaluation, enum
Other on-going IRT research

- ad-hoc mobile data exchange: DS7
- adaptive reservation and billing: RNAP
- resource reservations: YESSIR, BGRP
Summary

- complete architecture and set of components for building SIP parts of
  - VoIP ASP ("IP Centrex")
  - IP PBX
  - carrier VoIP network
- standards-compliant and cross-platform
- independently deployable, but common code base and administration