

# 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](mailto:schulzrinne@cs.columbia.edu)

January 2001

## Overview

---

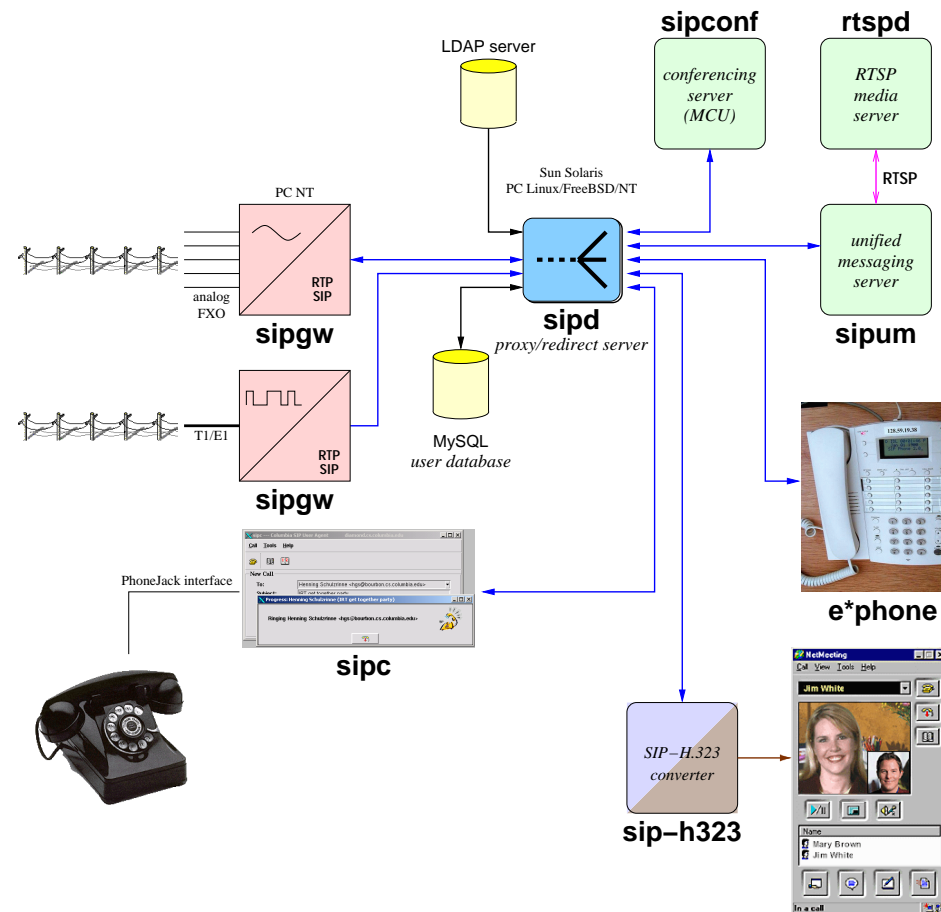
- architecture
- SIP library: C/C++ and Java
- SIP clients
- SIP proxy/redirect/application server
- SIP applications: unified messaging, conferencing
- related work: QoS, charging, mobility, 911 services

## 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

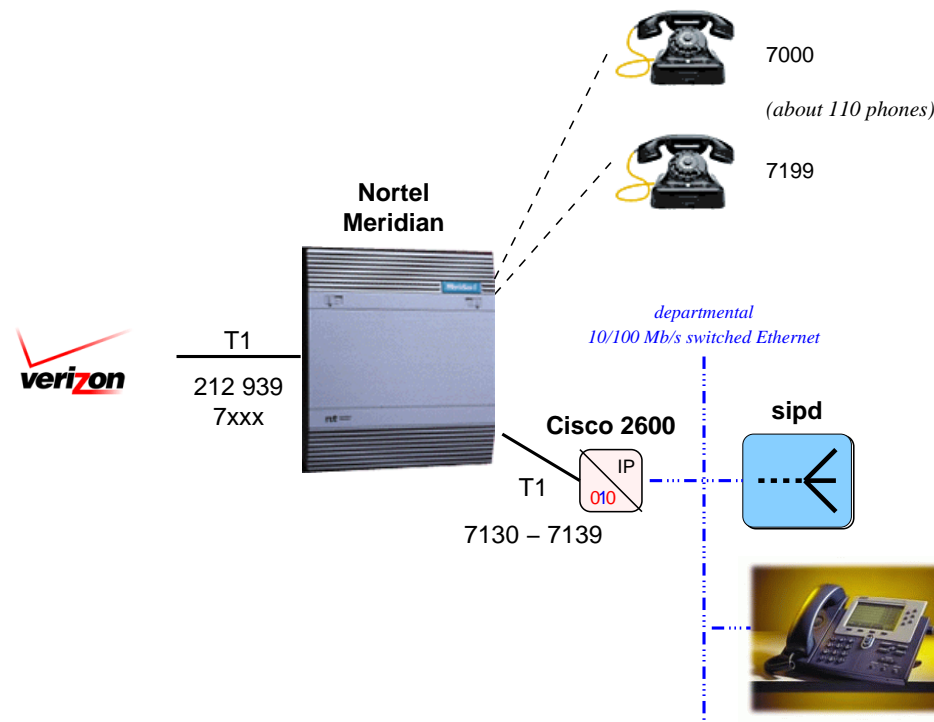
# Overall architecture



common code base for everything except sipc and e\*phone

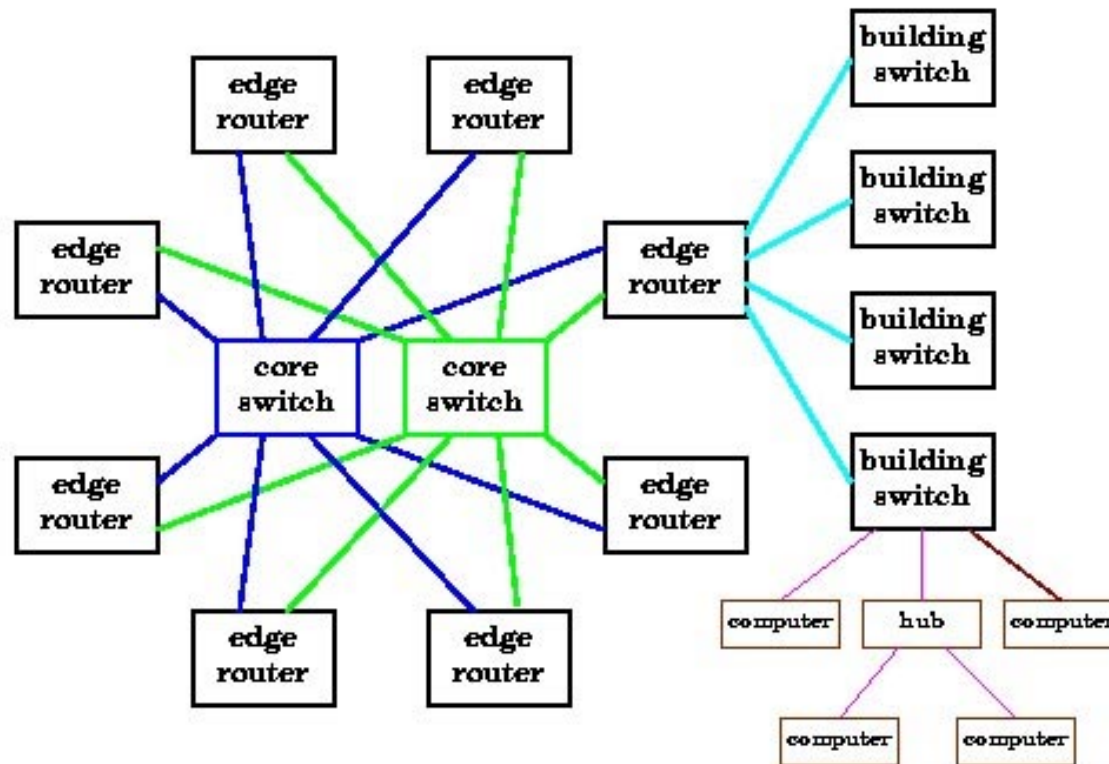
## Columbia University Computer Science test bed

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



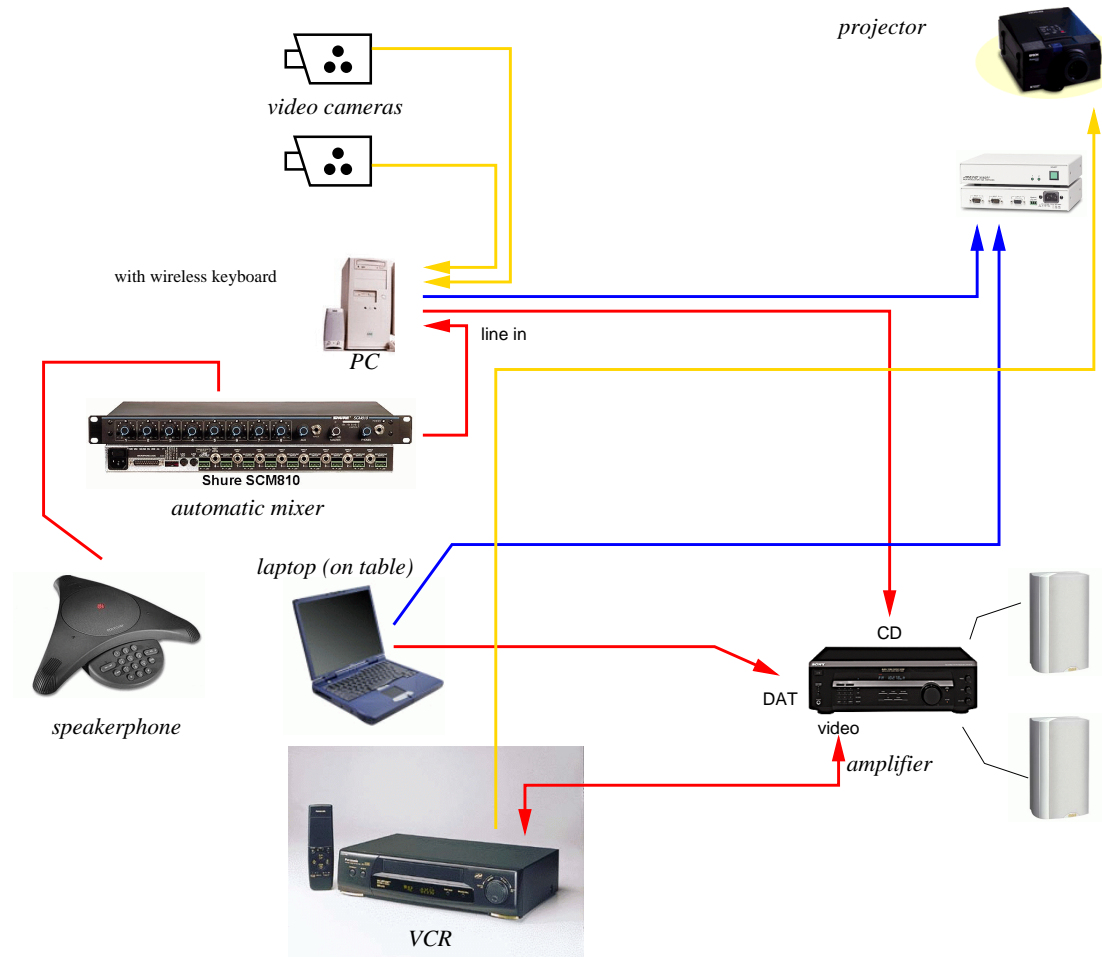
## Campus network

Cisco-based (6509) gigabit Ethernet:

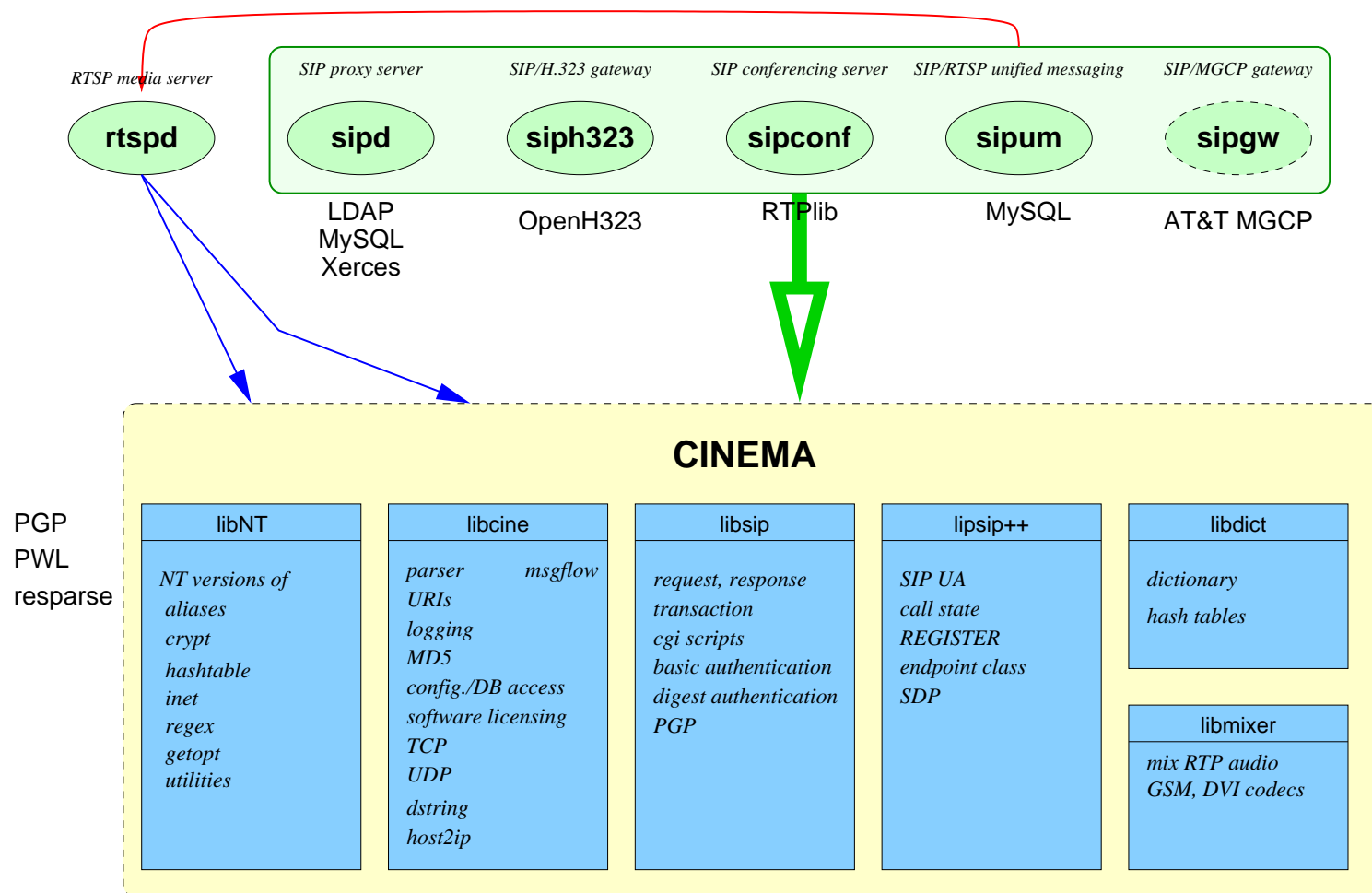


At least 4 switched jacks in each office.

# Columbia University CS conference room



# CINEMA





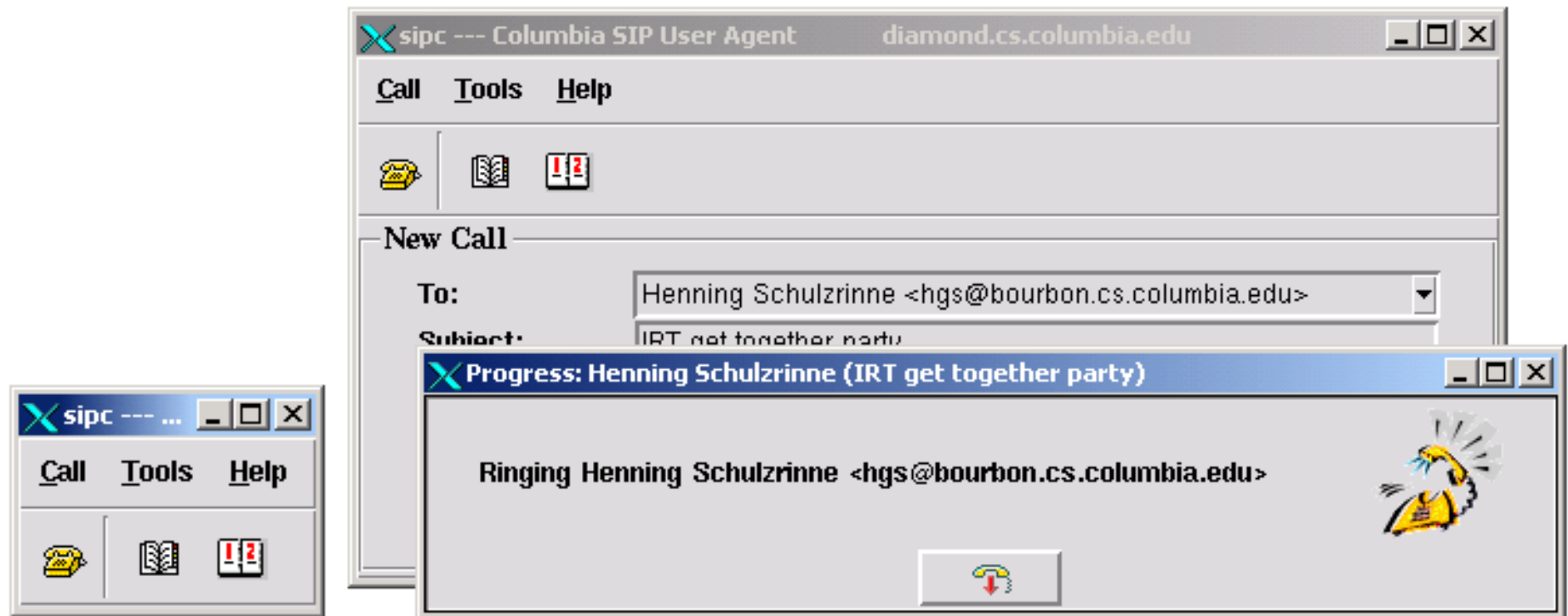
# e\*phone

DSP-based, single-processor Ethernet phone; being commercialized

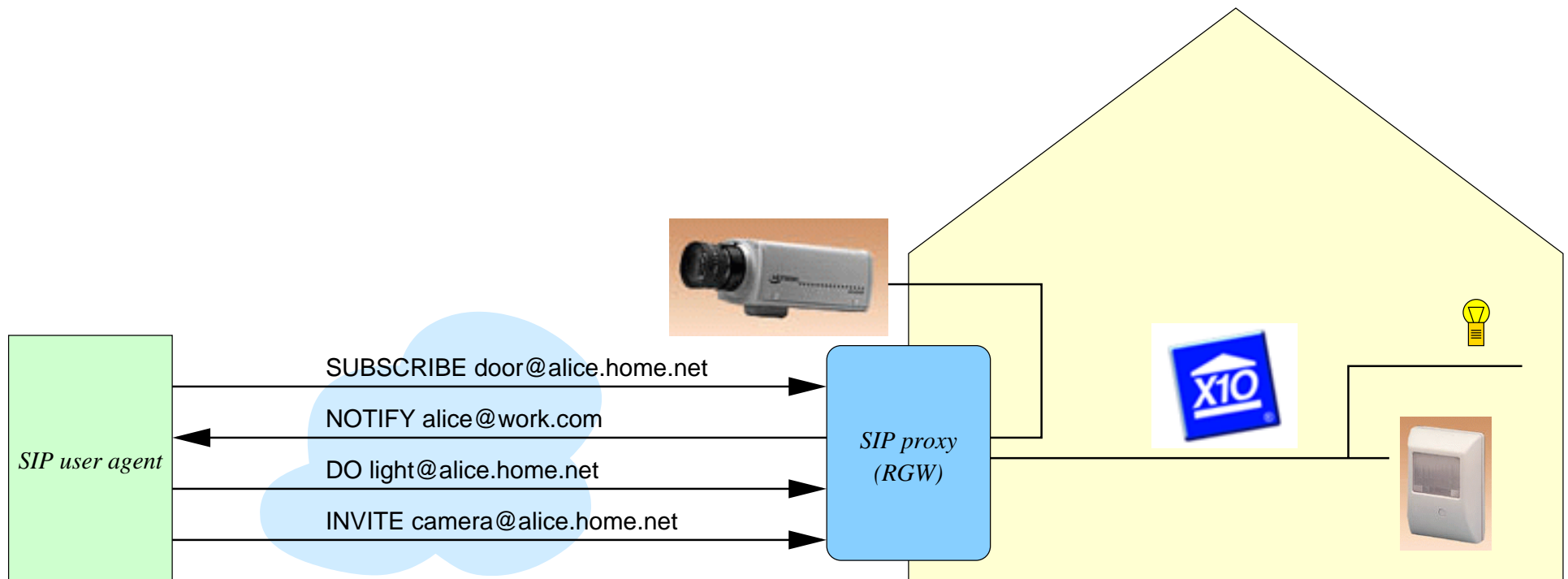


## SIP user agent sipc

---



## Device control



## 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

## CINEMA Registration

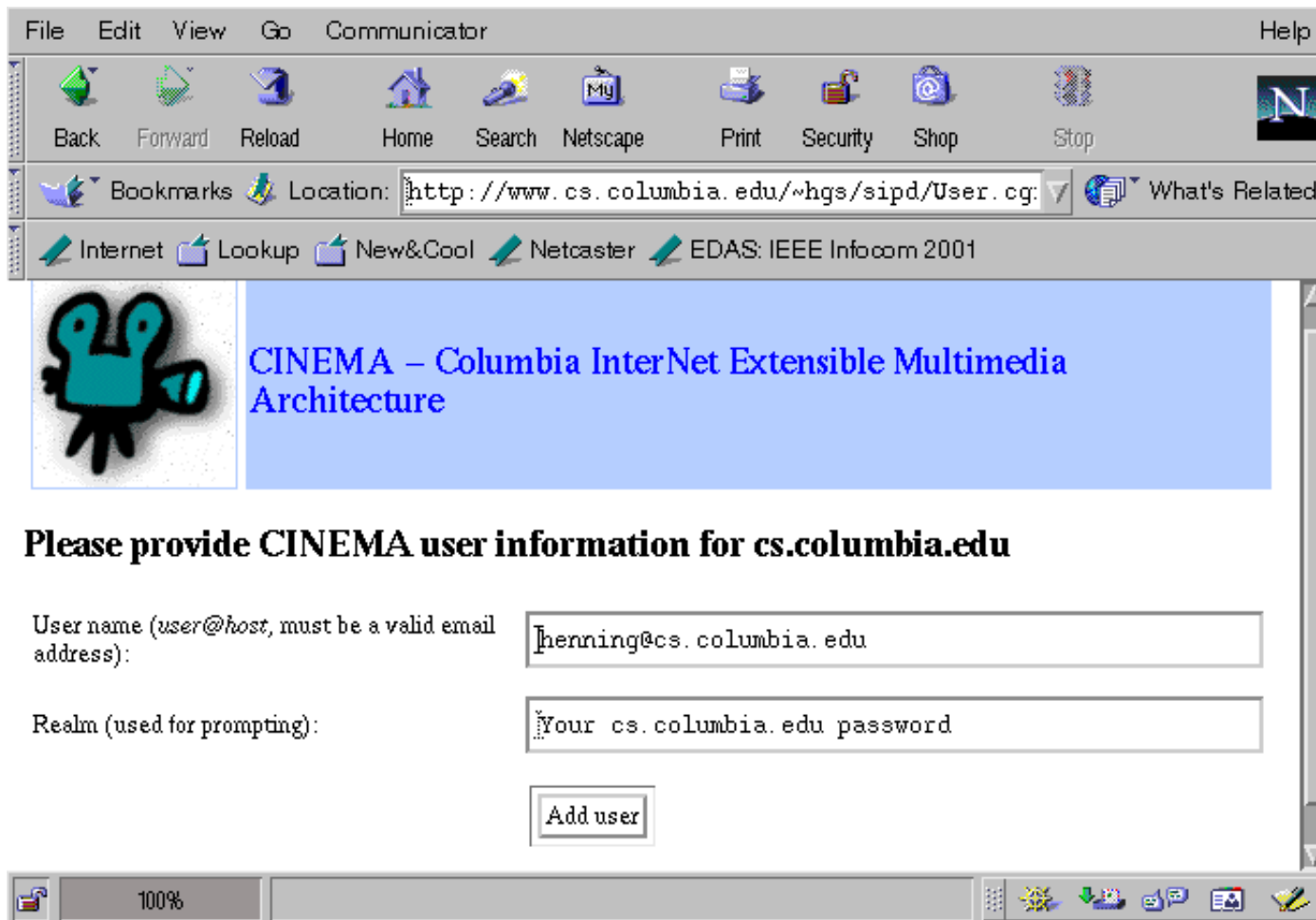
---

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



# sipd user configuration

The screenshot shows a Netscape Communicator browser window displaying the SIP User List configuration page. The browser's address bar shows the URL `http://www.cs.columbia.edu/~hgs/sipd/UserList.cgi`. The page features a logo of a blue crab and the text "CINEMA – Columbia InterNet Extensible Multimedia Architecture". Below the logo, the title "SIP User List" is displayed, followed by a note: "The information for `default@domain` is used as the default template for new users."

The main content is a table listing user configurations. The table has the following columns: User name (Click to edit), Realm, Groups, Authentication, Algorithm, SIP methods, Aliases, Contacts, Delete?, Users that can register for this user, and Last modified.

User name (Click to edit)	Realm	Groups	Authentication	Algorithm	SIP methods	Aliases	Contacts	Delete?	Users that can register for this user	Last modified
<a href="#">default@cs.columbia.edu</a>	Password for cs.columbia.edu	cgi voicemail	request	MD5	REGISTER INVITE			 		12 Oct 2000 18:43
<a href="#">henning@cs.columbia.edu</a>	Password for cs.columbia.edu	cgi voicemail	request	MD5	REGISTER INVITE			 		24 Oct 2000 21:48
<a href="#">hgs@cs.columbia.edu</a>	cs.columbia.edu	cgi voicemail admin	request	MD5	REGISTER INVITE	7042@cs.columbia.edu	<a href="mailto:hgs@cs.columbia.edu">mailto:hgs@cs.columbia.edu</a> <a href="mailto:hgs@erlang.cs.columbia.edu">hgs@erlang.cs.columbia.edu</a>	 	<a href="#">kns10@cs.columbia.edu</a> <a href="#">tk358@cs.columbia.edu</a> <a href="#">lennox@cs.columbia.edu</a>	12 Oct 2000 18:43
<a href="#">kns10@cs.columbia.edu</a>	Your login for kns10	cgi voicemail admin	required	MD5	REGISTER INVITE			 		12 Oct 2000 18:43
<a href="#">lennox@cs.columbia.edu</a>	Password for cs.columbia.edu	cgi voicemail admin	request	MD5	REGISTER INVITE			 		13 Oct 2000 11:11
<a href="#">schulzrinne@cs.columbia.edu</a>	Your cs.columbia.edu password							 		12 Oct 2000 18:42

## 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