Research status

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Internet Real-Time Lab

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Research interest
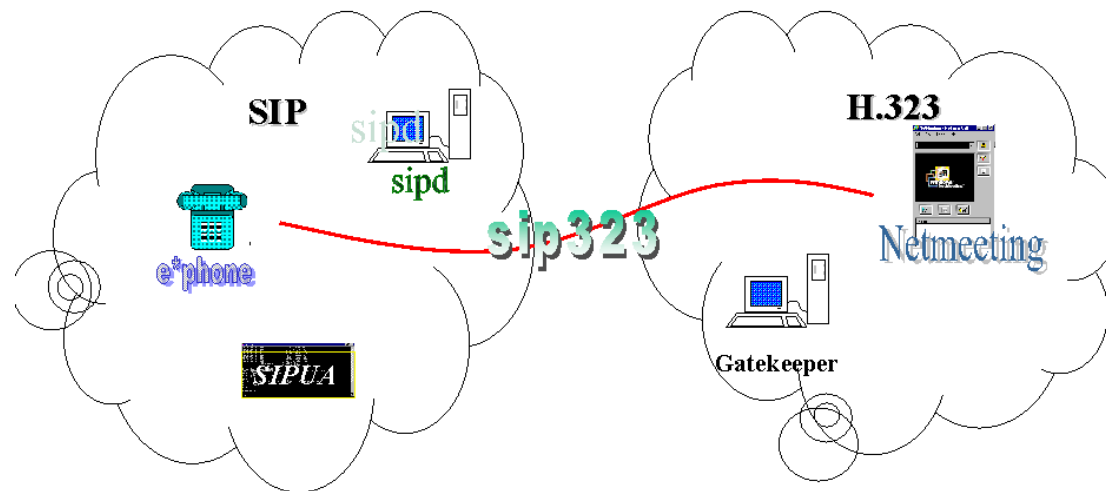
- Internet telephony

- SIP-H.323 interworking

- Unified messaging

IRT group meeting, Sep 14, 2000

Interworking between SIP/SDP and H.323

sip323 - http://www.cs.columbia.edu/~kns10/software/sip323

- Basic interworking draft, and paper

- Requirements document

- mailing list sip-h323@eGroups.com

- Simple point to point audio G.711 Mu law calls

- Built-in gatekeeper

- Different dialing modes

- . . .

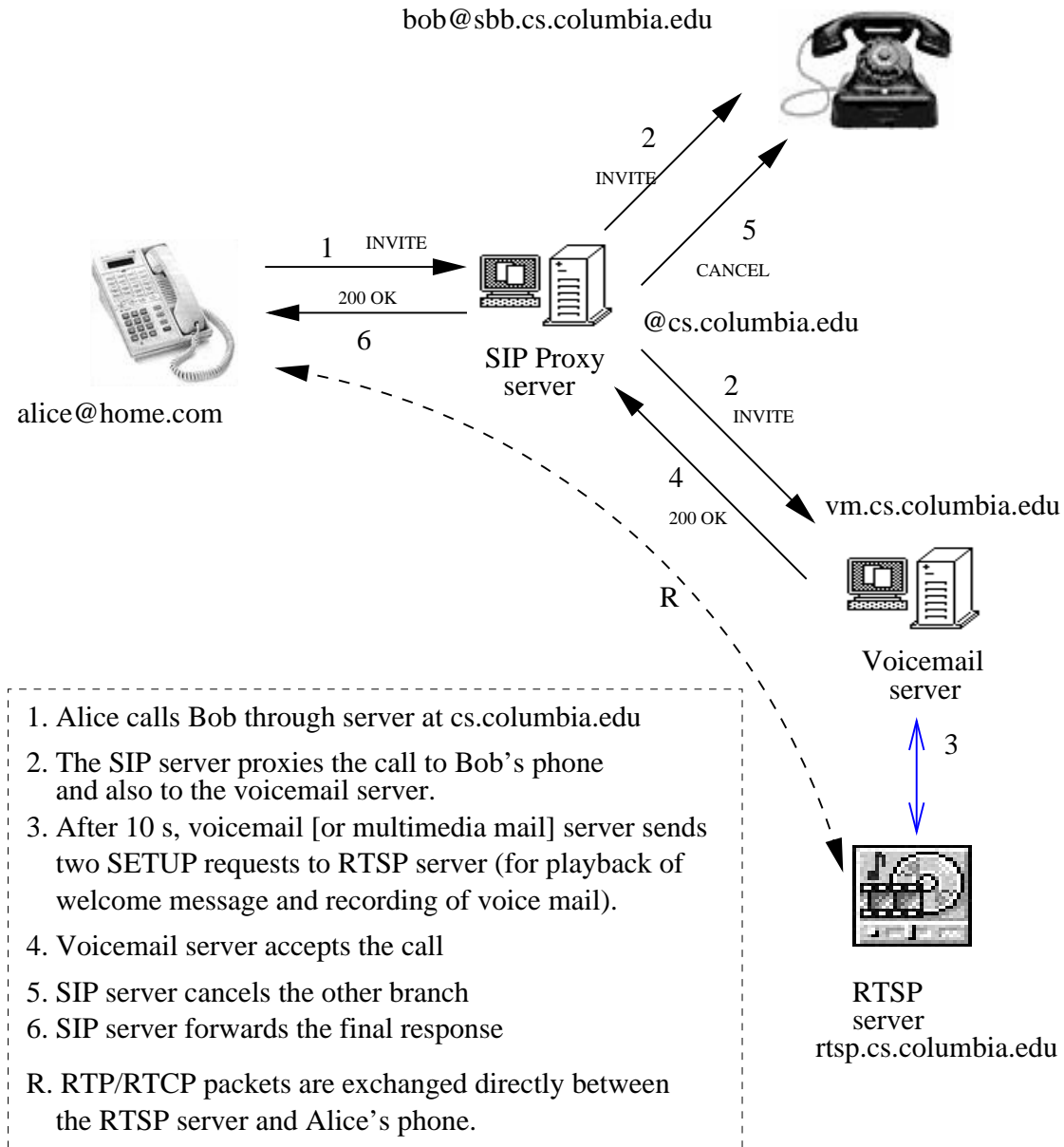# Interworking between SIP/SDP and H.323

SIP/RTSP Unified Messaging System

sipum - http://www.cs.columbia.edu/vmail

- Publication

- Message recording and playback

- Access from RTSP client, SIP useragent and Web browser

- Email notification

- Individual web based accounts

Future: common password file for sipd and sipum, per-user quota, integration of video, fax and email; deployment, . . .

bob@sbb.cs.columbia.edu

2
INVITE

5
CANCEL

1   INVITE

200 OK

6

SIP Proxy
server

@cs.columbia.edu

2
INVITE

alice@home.com

4
200 OK

vm.cs.columbia.edu

R

Voicemail
server

3

RTSP
server
rtsp.cs.columbia.edu

1. Alice calls Bob through server at cs.columbia.edu

2. The SIP server proxies the call to Bob's phone
and also to the voicemail server.

3. After 10 s, voicemail [or multimedia mail] server sends
two SETUP requests to RTSP server (for playback of
welcome message and recording of voice mail).

4. Voicemail server accepts the call

5. SIP server cancels the other branch

6. SIP server forwards the final response

R. RTP/RTCP packets are exchanged directly between
the RTSP server and Alice's phone.

sipum

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kns10 @voicemail.com

	Inbox		Folders		Options		Help

inbox

☐	►	Date	From		Subject	Size
☐		Fri 11, 10 AM	sip:hgs@cs.columbi...		Message: 215488.au	7 s (57 KB)
☐		Thu 10, 03 PM	sip:kundan@marta.c...		Message: 456212.au	4 s (39 KB)
☐		Wed 09, 11 AM	sip:333@216.66.67....		Message: 330307.au	22 s (182 KB)
☐	►	Jul 24	sip:user1@cs.colum...		test call	9 s (75 KB)
☐		Jul 24	sip:kns10@cs.colum...		[normal] Testing sipum/v-mail	5 s (42 KB)
☐		Jul 24	sip:kns10@cs.colum...		[normal] Demo call from sipc	9 s (77 KB)
☐	►	Jul 24	sip:kns10@cs.colum...		[normal] Demonstration call.	0 s (0 KB)

Delete Selected	-Move to Folder- ☐	Refresh	Change User	Permanent Signout

Forward selected mails to these email(s): I

Real Time Streaming Protocol

rtspd - http://www.cs.columbia.edu/h̃gs/rtspd

- Playback and recording of G.711 Mu law (.au files)

- Works with Apple QuickTime client

Future: Random positioning, multicast, live broadcast, other formats, . . .

SIP Conference Bridge

sipconf - http://www.cs.columbia.edu/~kns10/software/sipconf

- Multiple simultaneous conferences setup from web

- Audio mixing and play-out delay algorithm

- G.711 A law, Mu law, G.721, DVI ADPCM

Future: Video integration, additional codecs, authentication, time bound conferences, auto-registration, . . .
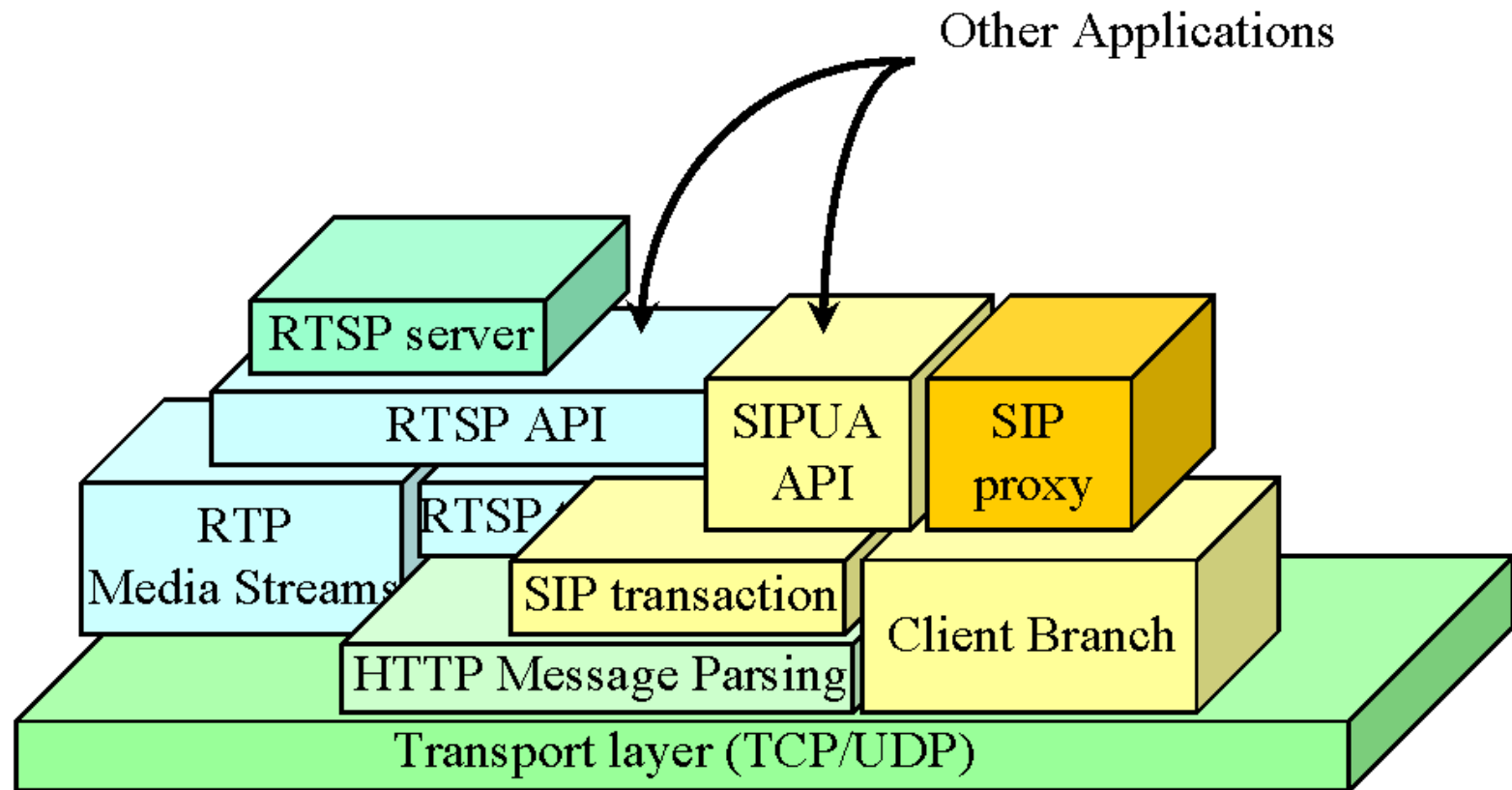
SIP library

libsip++ - http://www.cs.columbia.edu/~kns10/software/cinema

Goals

- Object oriented design (for C++ and Java)

- Extensible (RTSP, HTTP)

- Easy to use (sipum, sipconf, sip323, . . .)

- Modular and portable (solaris, windows, linux, freebsd, . . .)

Status

- Class library definition

- Documentation (design and API)

Architecture overview