Internet Telephony and Media-on-Demand Services

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Overview

- applications: telephone, TV, radio, . . .
- Internet integrated services architecture: RSVP alternatives
- real-time transport: RTP
- signaling: SIP
- stream control: RTSP

Applications

Internet Integrated Services Applications

- integrate ➠ single delivery mechanism:
  - telephony, multimedia conferencing
  - radio
  - cable television

Internet Telephony Advantages

- functional enhancements
- need LAN/WAN anyway
- (currently) not encumbered by regulatory burdens ("the FCC bypass")
- second phone ➠ reliability not as crucial
- no 7c/min access charge
- flat-rate billing
- cost of TAT transatlantic line is only $0.03/hour
- compression (but: already done for overseas)
- AT&T: "loose $350m in 2001 to Internet telephony"
- lower call-setup delay (H.324 with V.34: 14 seconds)

Internet Telephony Advantages: Control

- separation call and connection ➠ avoids triangle routing
- clear who handles in-band signaling (# sign: local switch, provider, called voice-mail system)
- no difference for call between intra-PBX, inter-LATA and general
- separation of connectivity from resource availability
- datagram ➠ less bootstrapping
- multiple call presences ➠ call waiting easier
- end system handles many functions
- fewer call states

Problems with Internet Telephony

- total DEM/minute: $5.5 \cdot 10^6$ (local + long-distance)
- ➠ total use 350 Gb/s, interstate 50 Gb/s
- "more than half is data"?
- predictable sound quality
- not suitable for dial-up users: signaling by email, telephone
- no advanced services (yet...)
- 640 mio. phone lines $\leftrightarrow$ 180 mio. computers ➠ interworking

Switching Costs

switching method	capacity (Gb/s)	cents/kb/s
10BaseT Ethernet hub	0.12	0.5
10BaseT Ethernet switch	0.24	2.0
100BaseTX Ethernet switch	0.80	1.0
router	2.1	16.0
local ATM switch	2.48	1.6
PBX (256 lines)	0.02	218.
AT&T 4ESS toll (100k lines)	6.40	7.8
AT&T 5ESS local (107k lines)	6.85	156.

Major cost is cabling + interface (both telephone and optical), not switching ➠ single infrastructure

Internet Telephony/Radio/TV Features

- adaptive compression: trade $ for quality
- flexible side channel (“caller id”, RMX)
- more than audio
- easy integration with email, WWW

Internet Radio

- 12,140 U.S. AM and FM radio stations, only 100 in Germany
- FM quality (56 kb/s) ➠ backbone capacity of 680 Mb/s
- New York City: 45 FM stations ➠ 2.5 Mb/s
- DirecTV: 31 audio channels ➠ 1.7 Mb/s
- easy time-shifting, content-labeling ➠ near media-on-demand
- niche markets ➠ “shortwave radio”

Daily Media Doses

media	minutes/day
long-distance calls, US	14
America Online (AOL)	32
local phone calls, US	38
total phone line usage	52
Internet	57
Television, Sweden	120
television, USA	240

Consumer Access

New service break $19.95 flat-rate model:

- acceptable RT bandwidth = acceptable web bandwidth
- 200 concurrent users / T1 ➠ 7.5 kb/s
- currently: 10 customers/modem (AOL: 30) ➠ will decrease
- ➠ only 75c/month for pipe ISP $\rightarrow$ backbone
- ➠ "Internet radio" costs $30/month for access

Internet Integrated Services Architecture

Internet Integrated Services Architecture

link-level: scheduling, link-layer-specific

IP multicast: trade-off state vs. tree optimality

reservation: set up scheduling state ➠ RSVP

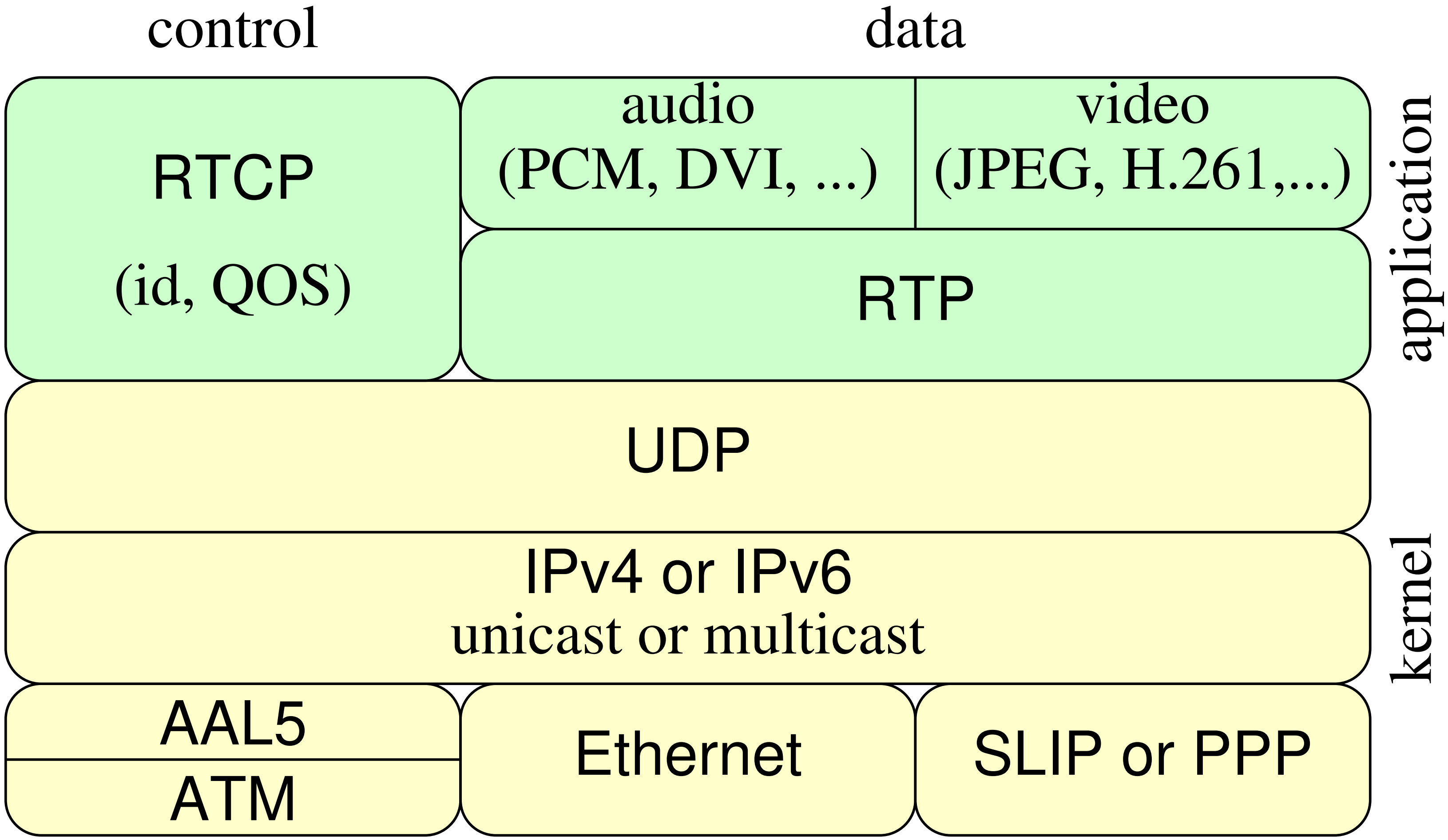
transport: timing, QOS feedback ➠ RTP

applications: playout delay adjustment, error concealment, bit-rate adaptation, VCR control (➠ RTSP), signaling (➠ SIP) ...

Internet Integrated Services Architecture

- philosophy: work both with and without resource reservation ➠ robustness
- most traffic remains best-effort ➠ reservation efficiency not too important
- few guaranteed service classes:
  - guaranteed load = “lightly loaded network”
  - guaranteed delay = Parekh-Gallagher bound, WFQ

Internet Multimedia Data Stack



+ RSVP (or TOS/precedence) when needed for differential service

RSVP Problems

- if reservation/tear down request lost, no immediate feedback
- can increase reservation latency or “phone off hook”
- large number of refreshes ➠ scaling problems
- hop-by-hop confirmation ➠ extend refresh interval
- receiver-oriented ➠ complexity (RESV, PATH), not always needed
- most applications don’t need full flow spec

Sender-Based Implicit Reservations

- appropriate for distribution applications, conferences
- use RTCP sender reports ➠ bandwidth
- soft-state: about 1% of data bandwidth
- IPv4, IPv6 router alert option ➠ forward packet to CPU
- requires no standardization
- error reports

Real-Time Transport Protocol

RTP

Data:

- end-to-end transport of streams with timing information
- labeling: source (independent of network source), payload type
- encryption, sequencing, loss detection, ...

Control = RTCP ➠ periodic ($T = f(N)$) reports to multicast group:

- member names:
  - member list
  - loop detection
  - multiple media $\rightarrow$ single conference
- sender reports ➠ rate estimation, lip-sync
- receiver reports ➠ loss, jitter, delay feedback

RTP Scaling

RTCP Algorithm

- RTCP packets sent periodically
- in $[1.25\text{s}, 3.75\text{s}]$ after joining
- media bandwidth b, RTCP packet size p, session size N
- period

$$T = \max(5\text{s}, \frac{pN}{0.05b \cdot 0.75})$$

- compute $\rightarrow$ set timer $\xrightarrow{T}$ send
- randomized $[0.5, 1.5]$ $\leftrightarrow$ packet synchronization

RTCP Scaling Problems

works fine for high-speed networks, smaller groups (< 1000)

initial transient: lots join at the same time: "movie at ten", channel surfing ➠ flooding, announcements lost

state storage: each receiver stores whole receiver list

slow estimate: learn only $\leq 1/\mathrm{s}$ for 28.8 kb/s

QOS transients: can't distinguish 50% loss, 1 minute vs. 5%, 10 minutes

multicast: DVMRP prune timeout ➠ RTCP packets flooded everywhere

BYE: everybody leaves at once

RTCP Transient Suppression

Conditional reconsideration: if N at send instant $> N$ at timer computation, recompute T and check interval

Unconditional reconsideration: recompute even if N hasn't increased

SendMax: accelerate convergence for new members ➠

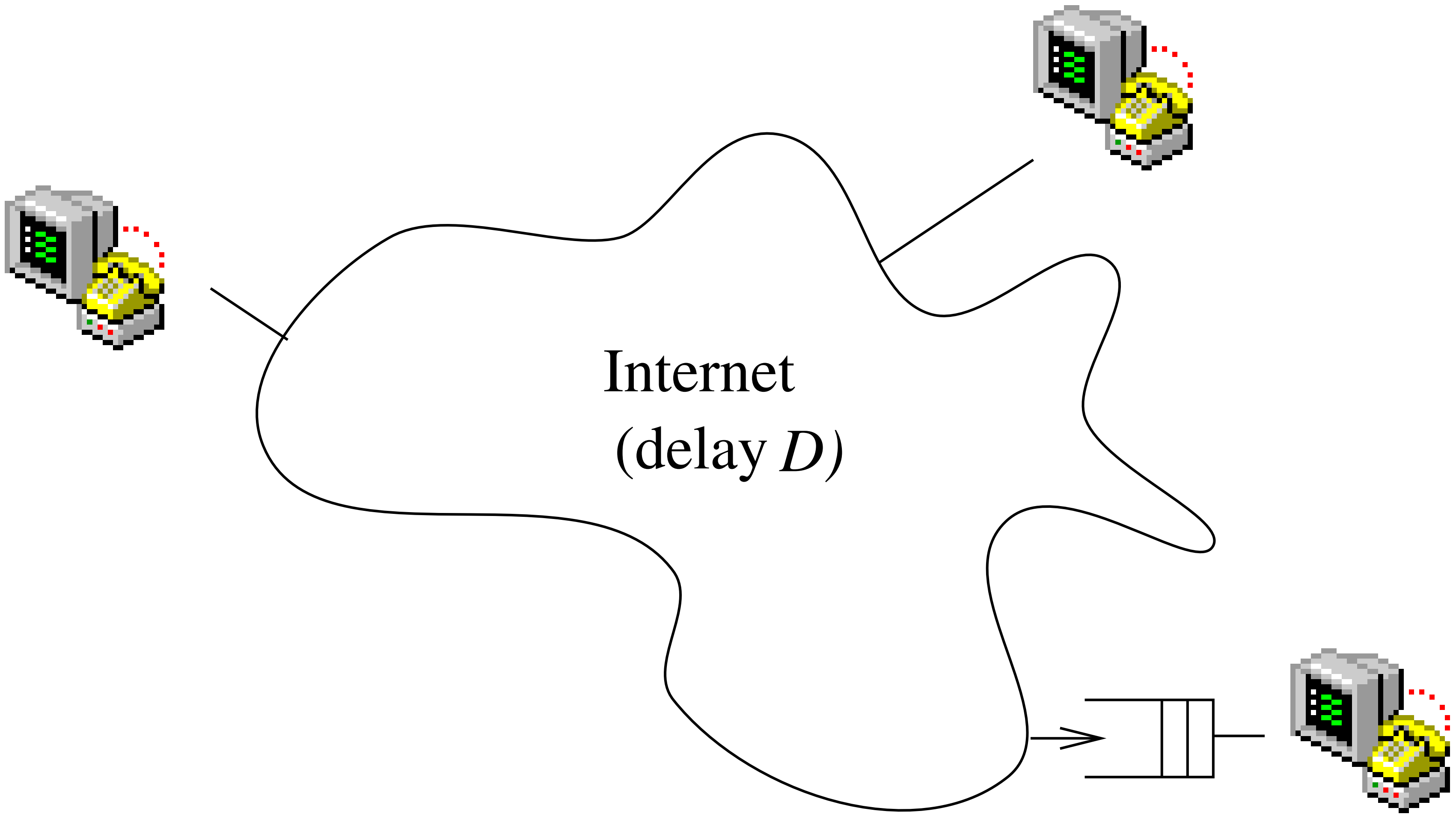
- send in RTCP packet: observed number of users
- compute interval $T = f(\max(\text{observed}, \text{allannounced}))$
- must track announcement for each member

Characteristics:

Learning curve: $L(t)$ = estimate at each member

Packets sent: ideally (no loss): = $L(t)$

Reconsideration: Network Model

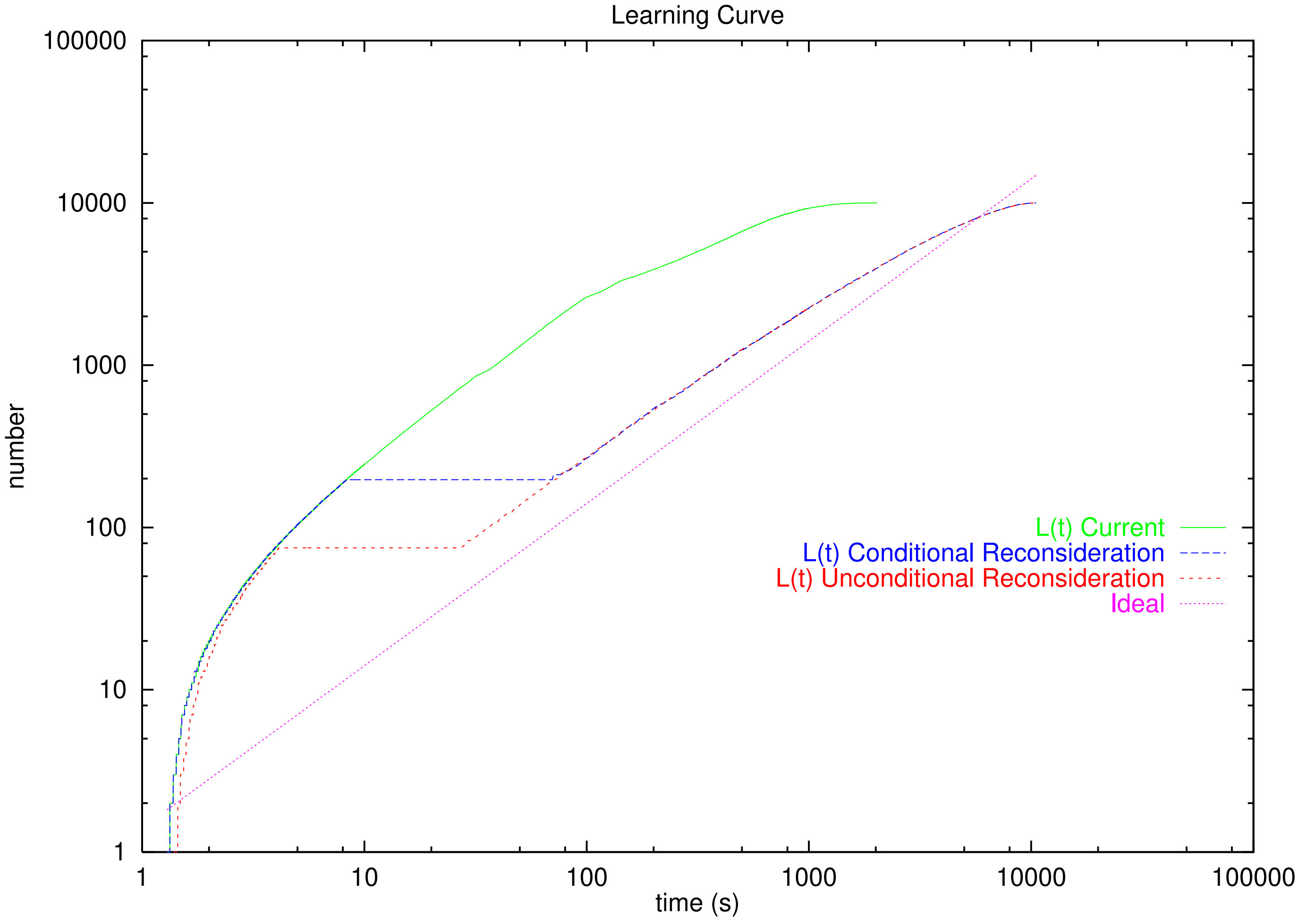


Reconsideration: Experiments

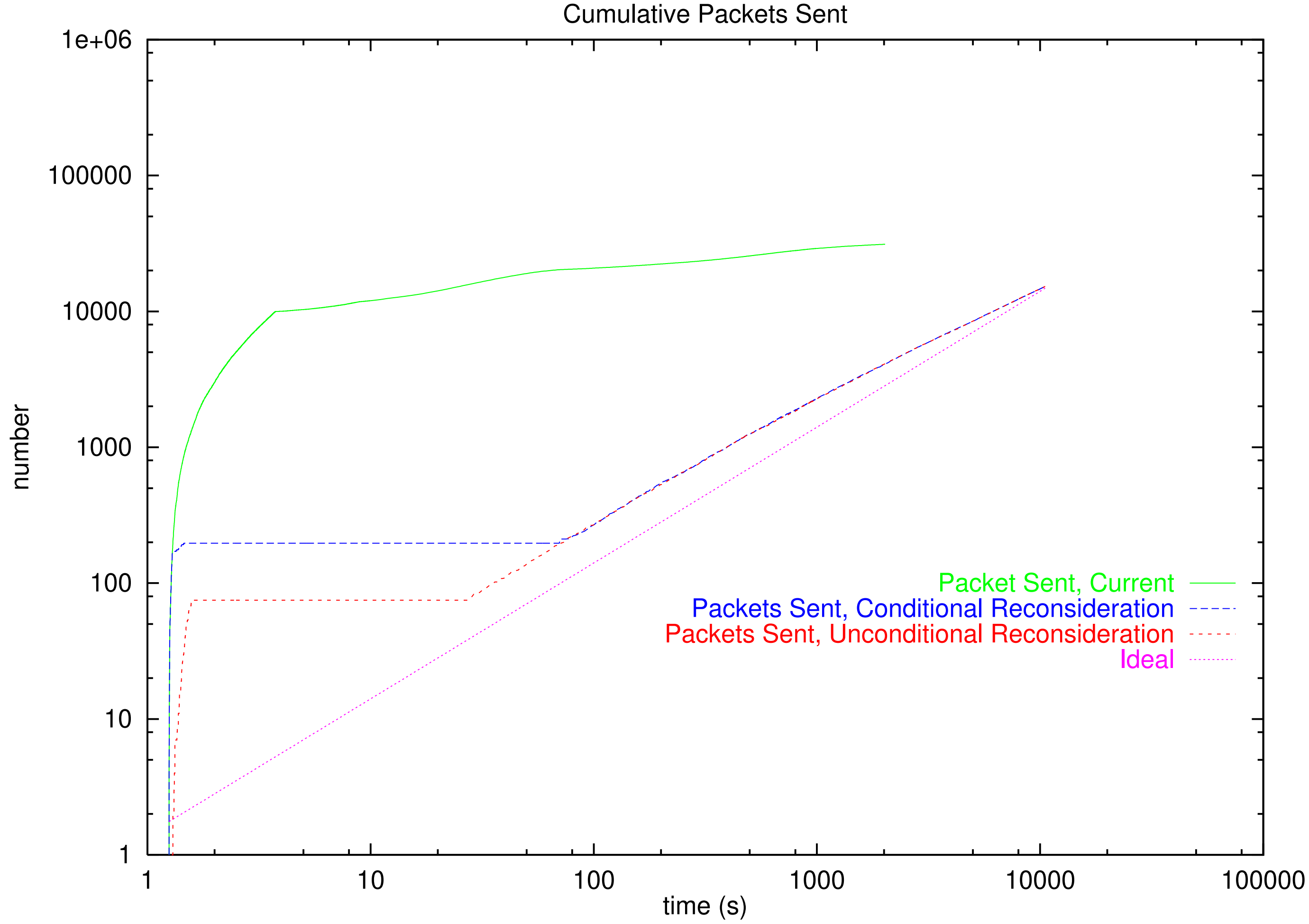
- 10,000 modem (28.8 kb/s) users in group
- join at $t = 0$ or 200 users every 200 ms
- 300 ms average delay, uniformly on [0 ms, 600 ms]
- 100 kB network buffer
- 128 byte RTCP packets

Note: backwards-compatible with existing spec

Reconsideration: Step Join



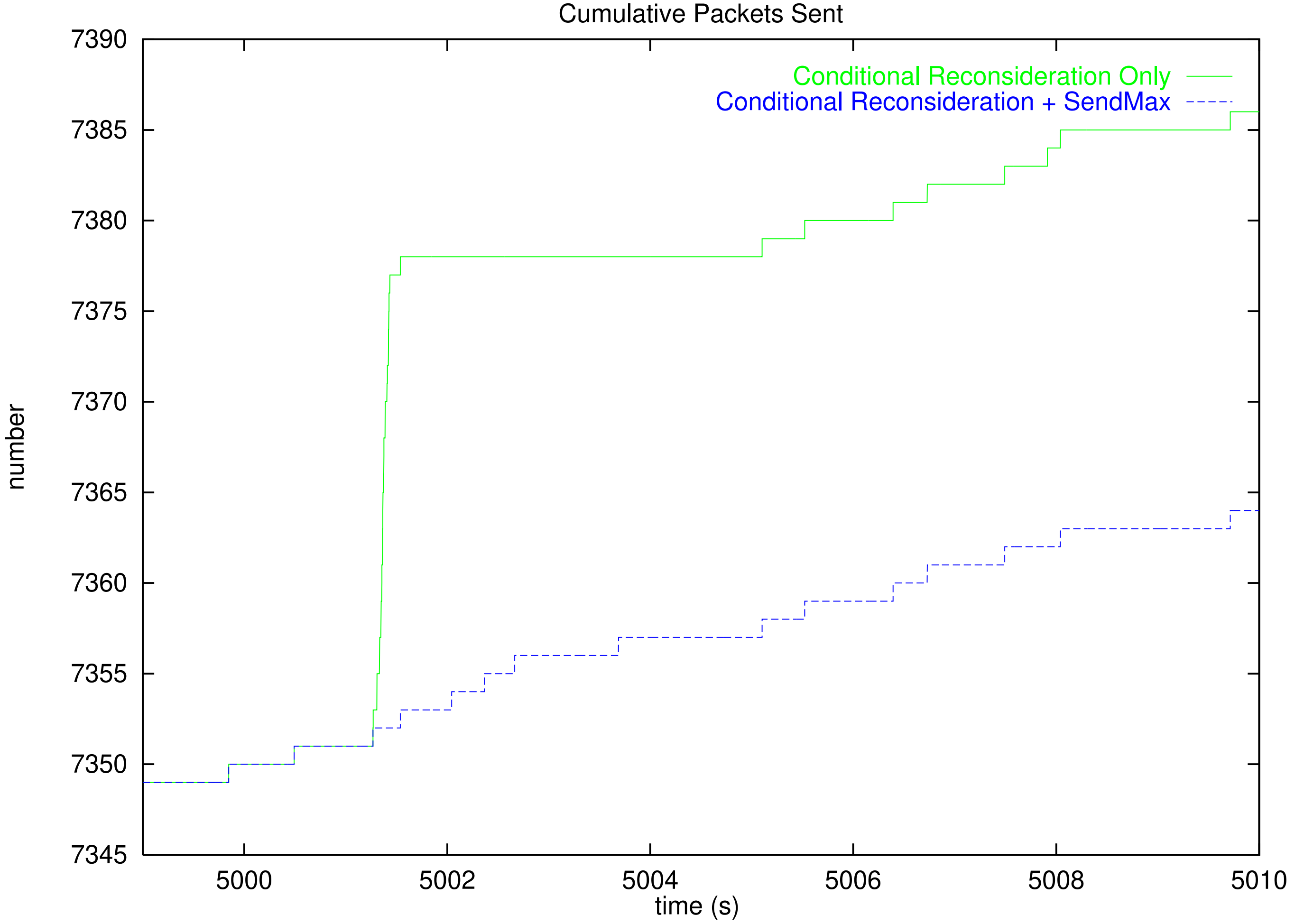
Reconsideration: Step Join



delay ➠ worse; slight dependence on distribution

Reconsideration: Linear Join

$t = 0$: +5000, $t = 5000$: +5000

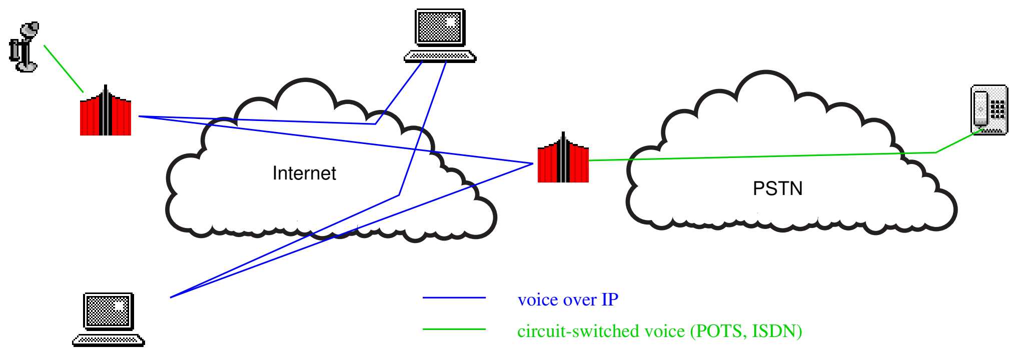


Internet Telephony Signaling

Internet Telephony Modes

Connectivity:

- packets end-to-end: Mbone, NetMeeting, ... ➠ requires continuous Internet connectivity
- PBX: use T1/PRI to server, packets on LAN
- tail-end hop off: use PSTN from POP



Bundling:

"Retail": individual packets per connection

"Wholesale" (trunking): interconnect PBXs or gateways

Signaling Needs

Internet: separate protocols for separate functions ➠

routing OSPF, BGP, . . . √

quality of service: RSVP √

user location: mapping identifier or name to location (IP address)

call set-up/teardown:

- call forwarding: one destination, try several, to answering machine
- call rejection with/without message
- parameter negotiation: media types, encodings, . . .

Internet Conference Model

- no central server(s) or multipoint control units
- hard to maintain consistent view of membership without central authority
- global view usually not necessary ➠ mostly cooperative participants
- "admission control" through encryption
- member $\in$ audio/video/…session $\in$ conference
- sessions $\equiv$ multicast group and port
- member $\equiv$ RTP CNAME (user@host) ➠ identification across sessions

Scenarios

1. call up another user ➠ “classical” telephony
2. invite new member to multicast conference
3. media-on-demand (unicast)
4. join on-going “broadcast”
5. invite media server or recorder to conference
6. device control: pan camera, adjust volume, ...

H.323

- H.323 = Q.931 (ISDN signaling) + H.245 (capabilities) over TCP
- very complex (200+ pages)
- no multicast signaling
- no multicasst conferences (➠ MCUs)
- no redirection
- but: capability negotiation (H.245)
- limited media instances (one video, audio stream)
- no media servers
- but: industry support…

Signaling Roles

- communications mode (phone, fax, mobile, pager, ...)
- negotiation of media and their parameters within mode
- resource reservation
- human accessibility

Ordering?

Session Initiation Protocol (SIP)

- don’t need to know IP address of user’s host
- many dial-up users do not have a permanent IP address
- invite to new and on-going conferences
- allow for manual and automatic forwarding
- *personal mobility* (complements data link/IP mobility) ➠ change of terminal (PC, digital cordless, palmtop), location
- call center: reach first (load distribution) or reach all (department conference)
- possibly change permissible encodings during conference
- integrate media-on-demand servers and recorders
- separate issue: QoS ➠ RSVP (but: provide parameters)

Naming

feature	phone #	email
mnemonic	no (except 1-800-Flowers)	name + org.
multiple	no	easy
characters	≈ 12	22
location-independent	1-700	yes: j.doe@ieee.org
carrier $\neq$ naming	maybe	yes
directory	411, 1-555, switchboard.com	haphazard

Could also use URL or URN, but no established user naming

Universal Communication Identifier

email address: RFC 822 ➠ president@whitehouse.gov or Bill Clinton <president@whitehouse.gov> ➠ needs to be translated

user@host: user name at physical host

telephone number: E.164 ➠ +1.202.The.Pres or phone://+1.212.555.1212@gateway.net2phone.com

Protocol

similar to HTTP (and RTSP) ➠

- HTTP = "the Internet RPC protocol"
- simple servers are easy, Apache for industrial-strength
- re-use HTTP extensions:
  - authentication
  - PICS = content labeling
  - JEPI = electronic payments
  - PEP = protocol extensions

but: also UDP (multicast!), bidirectional

Basic Operation

Example: call `schulzrinne@cs.columbia.edu`

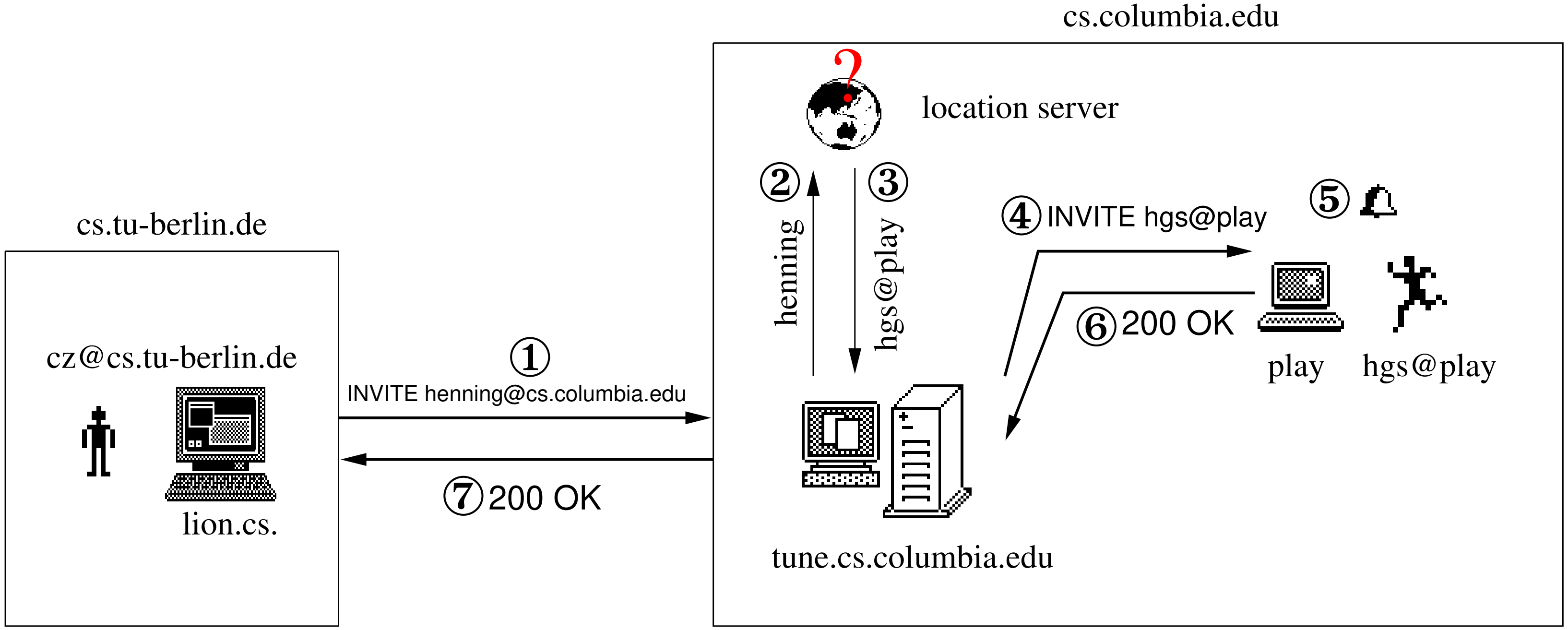
- may use directory or user location service (e.g., LDAP)
- caller resolves `cs.columbia.edu` to actual host using DNS SRV or MX records
- send protocol (SIP) request to server
- server either ...

  1. accepts/rejects/forwards call, maybe after asking user (invisible to caller)
  2. bridges to PSTN
  3. locates user ➠ forwards call if not local ➠ redirect mode ➠ client tries again
  4. acts as client ➠ proxy mode

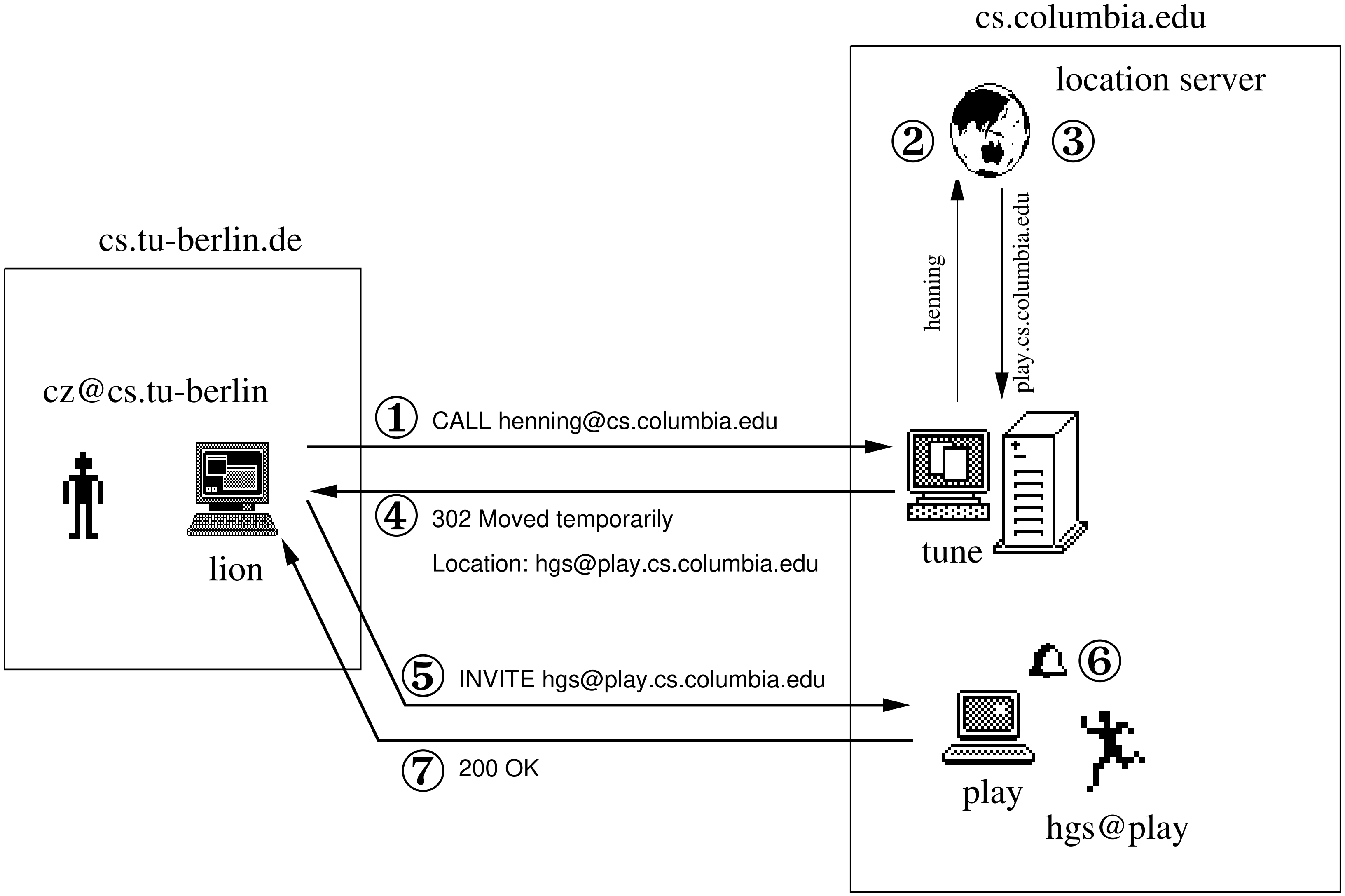
SIP: Interaction with User Location + Email

- user location is a local matter (but: could use SIP to register)
- if all else fails: use email EXPN or VRFY to get address ➠ `proud@alumni.columbia.edu`
- local server may use `.forward` information
- no SIP server found ➠ send MIME `application/sip` email
- integration with calendar manager ➠ information limiting

Operation in Proxy Mode



Operation in Redirect Mode



Protocol Request

```
INVITE schulzrinne@cs.columbia.edu SIP/2.0
From: Christian Zahl <cz@cs.tu-berlin.de>
To: Henning Schulzrinne <schulzrinne@cs.columbia.edu>
Via: SIP/2.0/UDP 131.215.131.131, SIP/2.0 foo.com
Session: 199612061103.AA1528@cloud9.cs.tu-berlin.de
Content-Type: application/sdp
Content-Length: 187
Subject: New error codes
```

session description

Response

- accept or reject call, manually or automatically
- redirect temporarily or permanently

Redirection ➠ indicate several possible locations

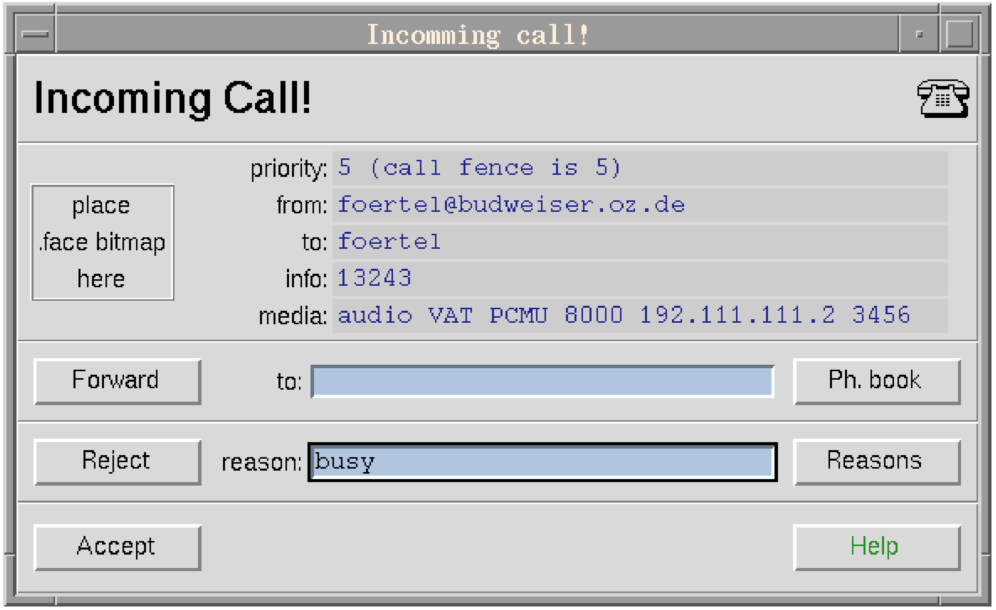
```
SIP/2.0 100 Trying to find user

SIP/2.0 150 Ringing

SIP/2.0 302 Callee has moved temporarily
Location: sip://jones@salt.lab3.company.com
Location: sip://jones@pepper.lab3.company.com
Location: phone://1.212.939.7042
```

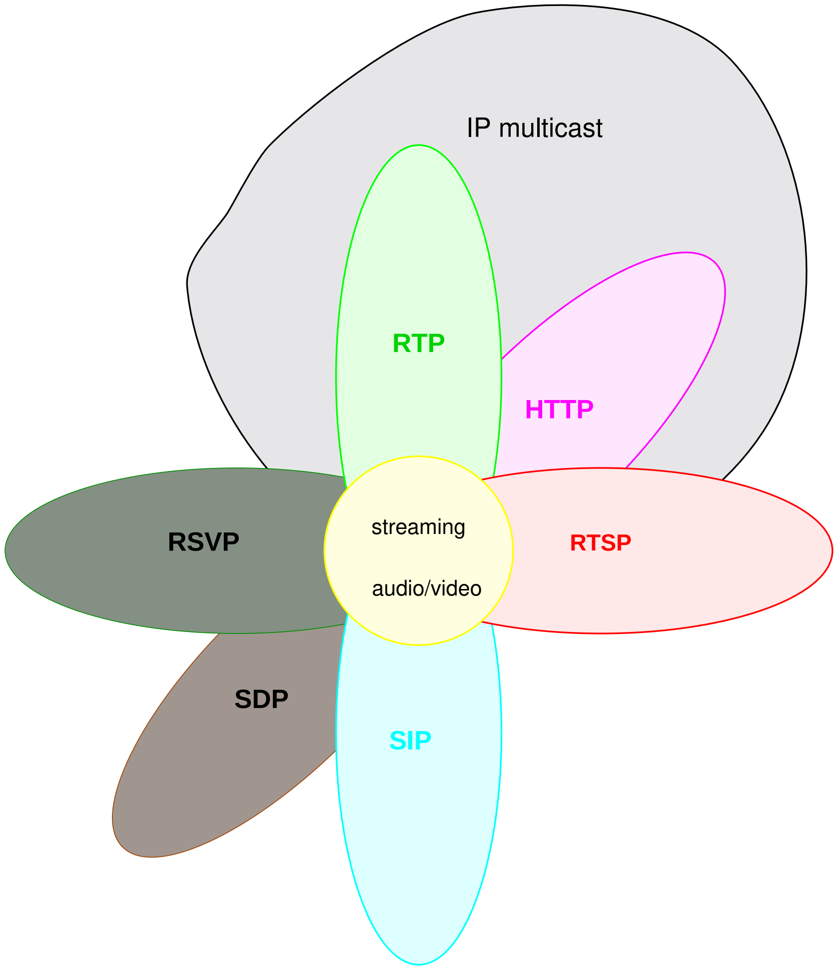
Implementation

- use SSL for security instead of separate signaling network
- implemented as PBX replacement
- reuse `.forward`, alias, mailcap mechanism
- part of PMM (conference bus) system
- configure media agents via WWW mailcap mechanism



RTSP – Real-Time Stream Protocol

Streaming Multimedia

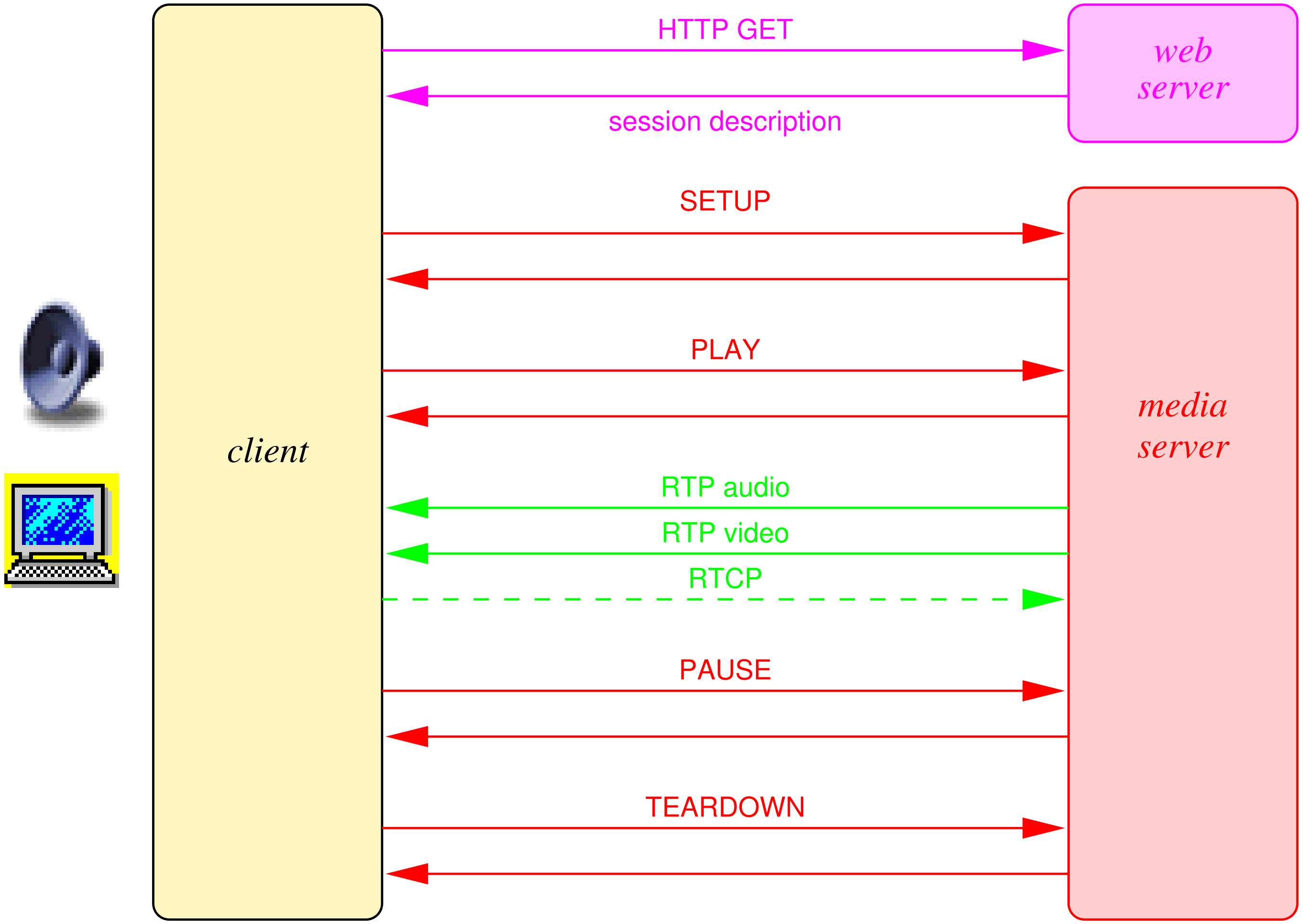


(similar functionality, much simpler than DSM-CC)

RTSP

- similar design as SIP (TCP + UDP, HTTP, . . .)
- TCP connection $\neq$ RTSP association
- “passing around the remote”
- connection/stream, connection/command, UDP
- session by identifier
- supports any session description
- control “tracks” (audio, video) and “presentation” (movie)
- remote digital editing

RTSP Operation



Open Issues

- Internet architecture and infrastructure
- reliability: “soft state” = robust, but not reliable
- charging and billing

Charging and Payments

- charging: multiple providers, multicast, congestion-based
- 800/888, 900 services
- model: flat rate for best effort, per-kb minute for reserved
- RSVP: include payment authorization in RESV packet
- with RSVP, all receiving parties pay (not the same?)
- prevent kickback schemes?
- include billing authorization for particular PATH msg id?

Summary

- price, features ➠ Internet as ISN
- data + control
- second (last?) chance to design telecom networks
- Internet signaling for telephony ➠ improved security and call handling
- media control and signaling closely related
- simple end-to-end signaling may replace CCS#7