Signaling for Internet Telephony Services

Henning Schulzrinne
Dept. of Computer Science
Columbia University
schulzrinne@cs.columbia.edu

OPENSIG Fall'96 Workshop
Columbia University

October 14-15, 1996

Overview

- Why Internet telephony?
- Modes
- Signaling functionality
- SCIP
- Open Issues

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 local (107k lines)	6.85	156.
AT&T 5ESS toll (100k lines)	6.40	7.8

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

Internet telephony advantages

- need LAN/WAN anyway
- 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/call

Internet telephony features

- adaptive compression: trade $ for quality
- low-volume
- source identification
- easier computer-telephony integration
- more than audio
- easy integration with email, WWW

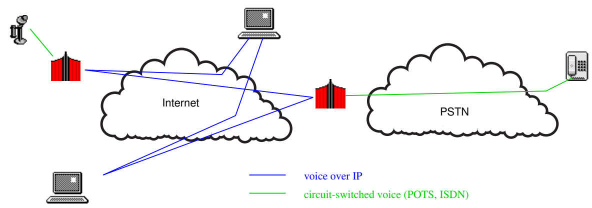
Problems with Internet telephony

- long-distance capacity: AT&T: 150 mio. calls/day, 3 min. each (?) ➠ 20 Gb/s
- predictable sound quality
- not suitable for dial-up users: signaling by email, telephone
- no advanced services
- 640 mio. phone lines $\leftrightarrow$ 180 mio. computers ➠ need interworking

Internet telephony modes

Connectivity:

- packets end-to-end: Mbone, Vocaltec, … ➠ requires continuous Internet connectivity
- tail-end hop off: use PSTN from POP



Bundling:

"Wholesale" (trunking): interconnect PBXs or gateways

"Retail": individual packets per connection

Signaling needs

Internet: separate protocols for separate functions ➠

routing OSPF, BGP, … $\surd$

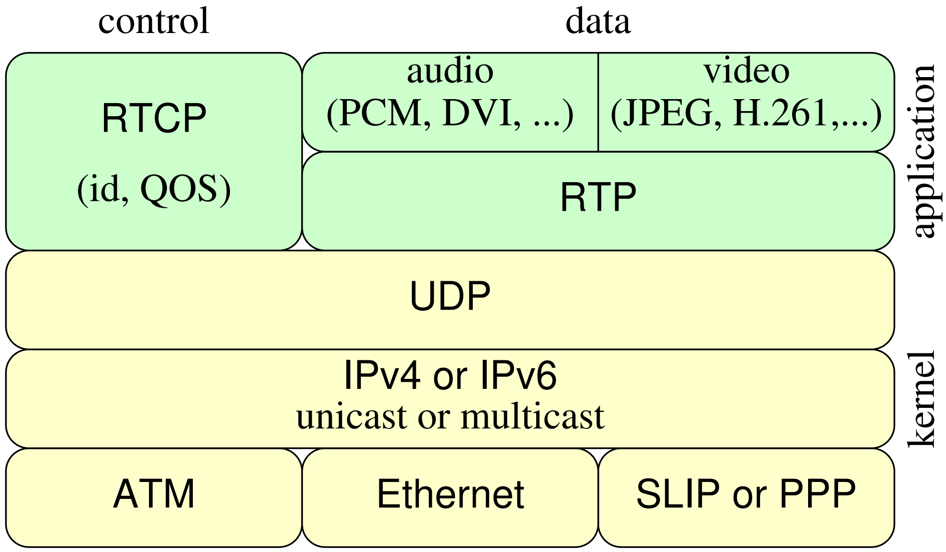
quality of service: RSVP $\surd$

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

call set-up/teardown:

- call forwarding: one, several, to answering machine
- call rejection
- parameter negotiation: media types, encodings, …

Internet multimedia stack



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

Conference invitation protocol: goals

- 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)
- reach first (load distribution) or reach all (department conference)
- possibly change permissible encodings during conference
- vehicle for conference control

Naming

feature	phone #	email
mnemonic	no (except 1-800-Flowers)	name + org.
multiple	no	easy
characters	$\approx$ 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, but no established user naming.

Universal Communication Identifier

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

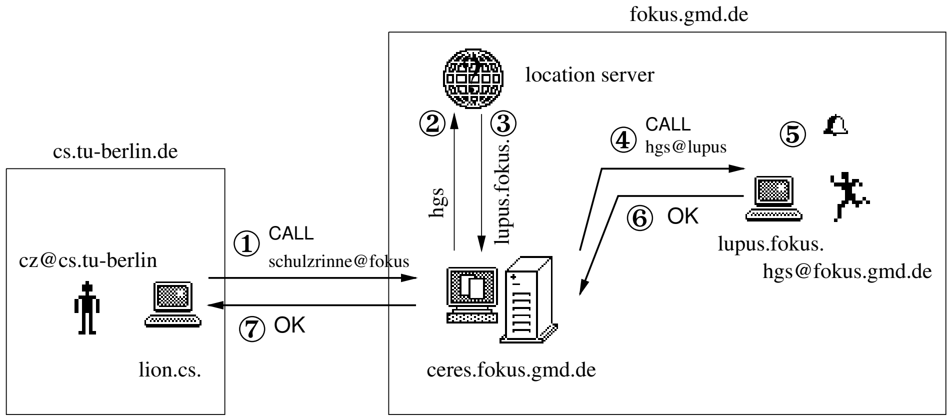
telephone number: E.164 ➠ +1.202.The.Pres or +1.212.555.1212@gateway.net2phone.com

Basic Operation

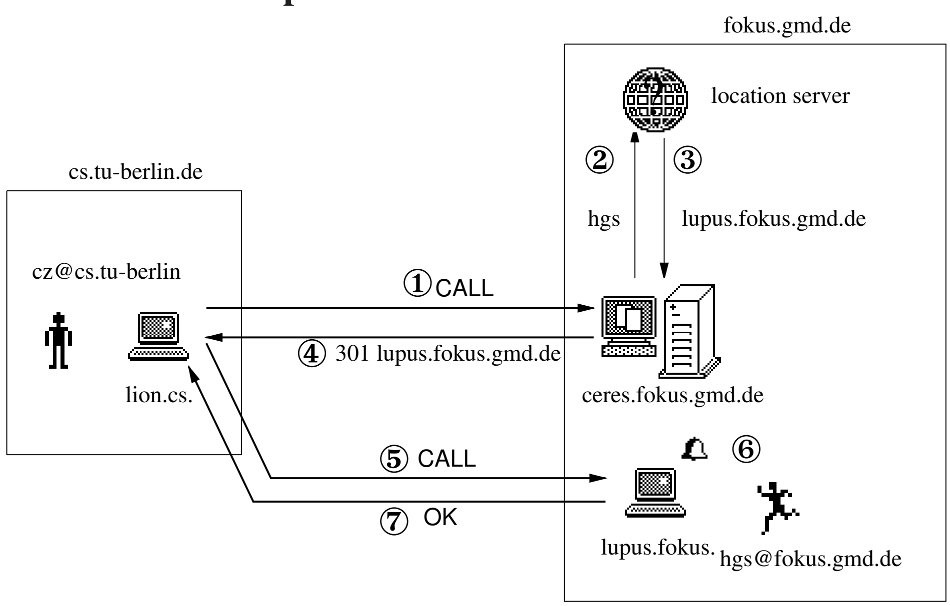
Example: call `schulzrinne@fokus.gmd.de`

- may use directory or user location service (e.g., LDAP)
- caller resolves `fokus.gmd.de` to actual host using DNS MX or SRV records
- send protocol (SCIP) 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
- user location is a local matter (but: could use SCIP to register)

Operation in proxy mode



Operation in redirect mode



Protocol

- similar to HTTP or SMTP: TCP-based, ASCII, client/server
- re-use HTTP authentication, encryption
- connection may stay open for change requests
- HTTP error codes, redirect
- disconnect through RTP BYE or timeout (soft-state)
- could use UDP

Protocol request

```
CALL hgs@lupus.fokus.gmd.de SCIP/1.0
From: Christian Zahl <cz@cs.tu-berlin.de>
To: Henning Schulzrinne <schulzrinne@fokus.gmd.de>
Call-Id: 9510021900.AA07734@lion.cs.tu-berlin.de
Referer: ceres.fokus.gmd.de
Expires: Mon, 02 Oct 1995 18:44:11 GMT
Phone: +1 413 555 1212
Email: Christian Zahl <cz@cs.tu-berlin.de>
Location: TU Berlin; tz=MET; loc=52 32 00 N 13 25 00 E
Priority: urgent
Content: application/mmsession
Reach: first
Subject: New error codes
```

session description

Response

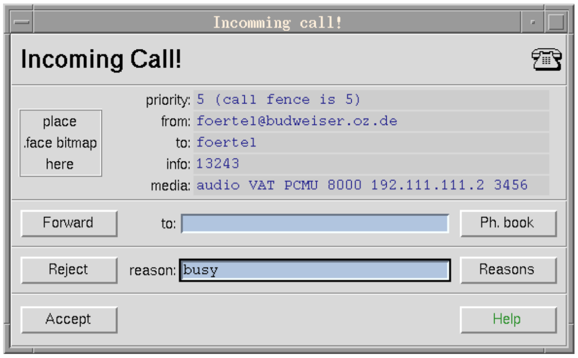
- accept or reject call, manually or automatically
- redirect temporarily or permanently

Redirection ➠ indicate several possible locations

```
SCIP/1.0 302 Callee has moved temporarily
Location: jones@salt.lab3.company.com
Location: jones@pepper.lab3.company.com
Location: +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



800/888 service

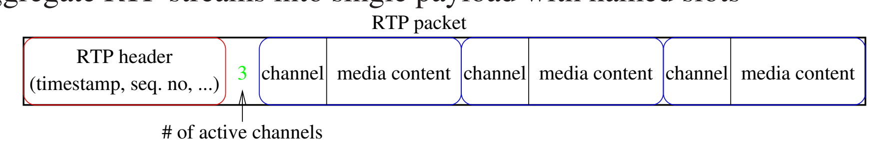
- Internet calls $\approx$ local calls
- has mapping capability ➠ 800/888 not needed for call distribution
- caller indicates charges ➠ callee accepts or declines
- opportunity for getting paid to call somebody
- pay using ecash ➠ caller not anonymous
- if reservation charges, make part of RSVP

900 service

- callee: `402 payment required`, indicates amount
- callee reconnects with ecash token
- if $/min, keep connection open?

Trunk signaling

- PBX ↔ PBX, gateway ↔ gateway
- aggregate RTP streams into single payload with named slots



- signal mapping of slots to destinations (UCI)

Future work

- trunk signaling details
- session specification
- call handling language
- interaction with lower-layer mobility
- extend this for streaming?
- integration with RSVP?

Summary

- price, features ➠ Internet telephony
- signaling = naming + mapping + connection
- Internet signaling ➠ improved security and call handling
- simple end-to-end signaling may replace CCS#7