Internet Telephony: Advanced Services

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Overview

- SIP servers and CO architecture
- authentication
- firewalls
- advanced services:
  - camp-on
  - hold
  - three-way calls
  - mobility
  - signaling compressed RTP
  - third-party control
- interfacing to ISDN
- billing

Role of SIP servers

one or more of...

- user location & call forwarding, hand-off to voice mail
- call processing language
- gateway to PSTN, ISDN, H.323
- call queueing
- advanced services: auto-dialer, operator services, ...
- authentication: add token to request
- service billing
- RSVP policy control?

Central office architecture

[role of BLR, SIP server, RSVP, ...]

SIP authentication

Basic: include plain-text password

Digest: challenge-response with shared secret

Certificate: sign non-Via parts of request headers, body with PGP, PKCS #7

SSL, SSH: but only for TCP

also need key exchange: Diffie-Hellman in SDP?

SIP firewalls

- act as standard SIP proxy
- much easier than H.323:
  - single protocol vs. H.225.0 + H.245
  - SDP $\ll$ H.245.0
  - single-stage negotiation
  - no need to maintain TCP connections during call
- need to understand INVITE, ACK and BYE
- if final SDP always also in success ACK, only ACK needed
- just parse INVITE SDP description if firewall limits incoming

Building advanced services

Construct from element *behavior*, not feature descriptions:

request URL: next resolution stage

From: logical call source

To: logical call destination

SDP "c=": address media is to be sent to – Internet or PSTN!

Also, Location: indication of additional/replacement participants

- SIP responses go to requestor
- don't ring for new additional participant in existing call ➠ call transfer
- BYE terminates From leg only
- call ends when last party leaves

Call disposition

Call-Disposition header specifies desired treatment of call:

all: reach all users in list

queue: queue call

do-not-forward: no proxying

...: extensible

Camp-on service

Choices:

1. callee indicates time to call back
2. "polling": caller issues repeated INVITE
3. caller indicates desire to wait:

```
C->S: INVITE sip://watson@example.com SIP/2.0
      Call-Disposition: queue

S->C: 181 Queued: 2 pending
      181 Queued: 1 pending
      200 OK
```

Hold

➠ temporarily disable media delivery

- multicast: use RTCP "interest indication"
- thus, unicast only
- send INVITE with SDP port number = 0 for media

music-on-hold ➠

- ask RTSP server to stream to callee address
- send INVITE with SDP address of music server (multicast!)

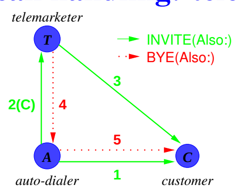
Outgoing call handling

Three-party setups:

- secretary dials for boss
- auto-dialer hands call to telemarketer
- attended call transfer
- operator services

➠ treat as three-party calls

Outgoing call handling: telemarketing



1. auto-dialer A INVITEs customer C
2. on answer, A INVITEs next available telemarketer T
3. A sends INVITE with Also: C to T
4. T INVITEs C for same call; C accepts automatically
5. T sends BYE to A
6. A sends BYE to C

SIP for application-layer mobility

- Internet telephony doesn't require full mobile IP functionality
- mobile terminal acquires local addresses via DHCP
- on move, SIP REGISTER to home base or regional LR
- maybe: include hints for next move (from GPS?)
- caller finds current location via SIP
- during call, either side sends INVITE with new address

RTP Header Compression

- IP + UDP + RTP headers = 40 bytes > payload for audio
- expensive for modem dialup lines
- compress whole stack, not just RTP
- but: most fields do not change from packet to packet, or change by a fixed constant (timestamps, sequence numbers)
- for each connection, remember constant fields and 1st order differences
- error indications in reverse direction upon failure
- ➠ 2 or 4 bytes header
- duplex, low-error point-to-point only ➠ access lines
- harmless (but useless) if not RTP
- does not apply to WAN connectivity – compresses IP

RTP Header Compression: SIP signaling

- modem access gateway or router needs to intercept RTP packets
- support limited
- can't easily detect RTP packets: no protocol ID, any port number
- fix: use SIP via proxy

Control of PSTN devices

- control local telephone switch ➠ end system issues requests
- "click-to-dial-back": web server issues requests

Outgoing calls:

```
INVITE sip://1-212-555-1234@net2phone.com
From: j.doe@example.com
Content-type: application/sdp

v=0
o=user1 53655765 2353687637 IN IP4 128.3.4.5
c=PSTN E.164 +1-212-939-7042
t=0 0
m=audio 0 RTP/AVP 0
```

Third-party control: incoming calls

action	SIP message
accept call	200
blind transfer	BYE, with new destination
forward, no answer	408
forward, busy	600
forward call	301 or 302, Location contains address

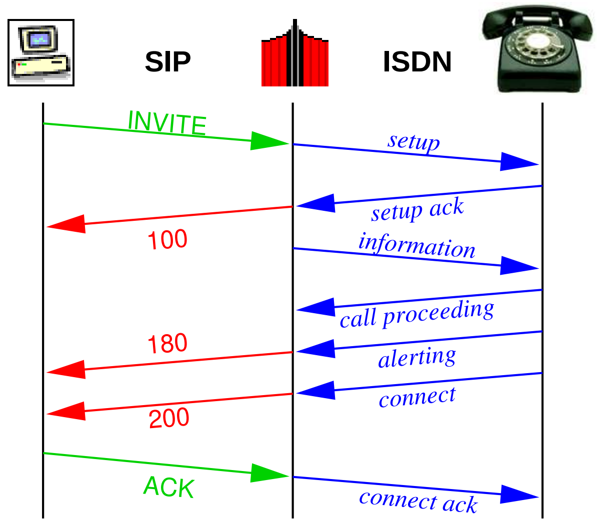
Configuring the PBX

```
REGISTER sip://pbx.example.com SIP/2.0
From: j.doe@example.com
Location: phone://1-212-939-7042
Content-Type: text/parameter

autoanswer: on
callsparked: 5
forward_busy: +1-415-555-1234
forward_all: off
```

returns current settings:

```
SIP/2.0 200 OK
Content-Type: text/parameter

music-on-hold: Alison's Restaurant
autoanswer: on
callsparked: 5
forward_busy: +1-415-555-1234
forward_all: off
```

SIP interaction with Q.931 (ISDN)



Billing

PSTN gateway services: RADIUS accounting (start/stop, bytes)

Internet resources: need RSVP policy protocol + billing aggregation

SIP services: SIP authentication + billing aggregation

ecash? credit-card interface?