SIP for PINT Services
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Services
- same from a network perspective:
  - control PBX
  - control SCN
- ➤ third-party call control
- click-to-dial = initiate outgoing call
- incoming call
- place calls on hold
- call transfer
- call parking
- conference calls
- install call handling functionality
**Additions/Changes to SIP and SDP**

- URLs in SIP can be:
  - `sip://user@host`
  - `sip://phone number@gateway`
  - `phone://phone number` for direct-dial
- no changes to SIP
- separation of logical call source/destination (SIP) and physical parties (SDP)
- SDP: add new address type “E.164”
- also works for H.2xx multimedia calls
- security mostly provided: SSL, SSH, S-HTTP, Authorization

**Architecture**

- Internet
- PSTN
- j.doe@isp.net
- +1 415 555 1200
- scn.example.com
- +1 212 555 1234
Outgoing Call

Request sent to scn.example.com:

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

v=0
o=user1 53655765 2353687637 IN IP4 128.3.4.5
c=PSTN E.164 +1-415-555-1200
t=0 0
m=audio 0 RTP/AVP 0
camp-on ➤ Call-Disposition: Queue

Answering Incoming Calls

Gateway sends INVITE to Internet host

accept call 200
blind transfer BYE, with new destination
forward, no anser 408
forward, busy 600
forward call 301 or 302, Location contains address
Placing Calls on Hold

- send INVITE for existing call-id
- SDP: media description port = 0
- media-specific hold

Call Parking

- send INVITE to gateway for existing call-id
- change SDP to indicate new destination (extension)
Call Transfer

- blind: \texttt{BYE} with "\texttt{Location: new phone number}"  
- supervised  
  1. place existing call on hold  
  2. place call to transfer destination and announce call  
  3. send \texttt{BYE} with "\texttt{Also: new address}"  

Group Calls: Internet  
Add a phone to an existing phone or Internet conference  
- several \texttt{INVITE}s from one or more end points  
- same session identifier in SDP = conference number  
- remove using \texttt{BYE}
Establishing Call Handling

- REGISTER
- call processing language
- many possible, to be defined by SIPTEL

REGISTER 7042@pbx.example.com SIP/2.0
Content-Type: application/cpl

state incoming {
    if {{string match *@insurance.com $caller}} {
        reject 303 -location sip://salesman@dial-a-joke.com
    }
    if {{clock hours} > 20} {
        reject 303 -retry-after tomorrow 9:00
    }
}

Configuring a PBX

REGISTER wit parameter list:

REGISTER 7042@pbx.example.com SIP/2.0
Content-Type: text/parameter

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