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# 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

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## 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

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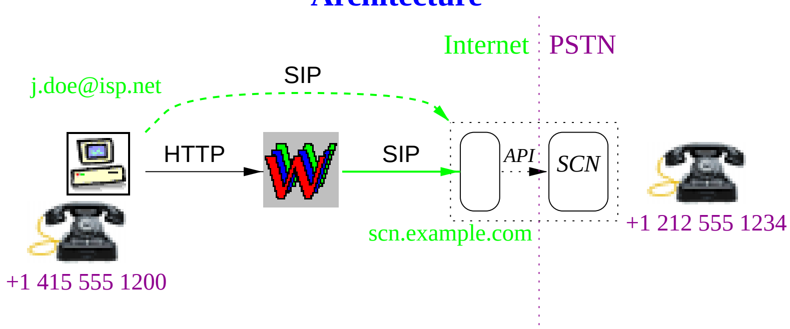
## Architecture



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## 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

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## 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 |

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## Placing Calls on Hold

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

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## Call Parking

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

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## Call Transfer

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

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## Group Calls: Internet

Add a phone to an existing phone or Internet conference

- several INVITEs from one or more end points
- same session identifier in SDP = conference number
- remove using BYE

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## 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
  }
}
```

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## 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
```