SIP Basics

CSG VoIP Workshop

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Outline

- What is SIP
- SIP system components
- SIP messages and responses
- SIP call flows
- SDP basics/CODECs
- SIP standards
- Questions and answers
But First…

Before we talk about VoIP let’s talk about systems and standards

The Electrical System

and predicting the future.
Western Electric Catalog c.1916

![Diagram of a flush receptacle](image)

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**“SPARTAN” STANDARD FLUSH RECEPTACLES—10 AMPERES, 250 VOLTS**

For combination plates (as defined elsewhere) specify “F” sections to accommodate these receptacles.

<table>
<thead>
<tr>
<th>No. 120</th>
<th>Inside</th>
<th>Outside</th>
</tr>
</thead>
<tbody>
<tr>
<td>KA-120 Body with standard cap</td>
<td>2 1/2</td>
<td>3 1/2</td>
</tr>
<tr>
<td>KB-120 Body with brass-covered cap</td>
<td>2 1/2</td>
<td>3 1/2</td>
</tr>
<tr>
<td>KC-120 Body with finger grip cap</td>
<td>2 1/4</td>
<td>3 1/4</td>
</tr>
<tr>
<td>KD-120 Body with elongated cap</td>
<td>2 1/4</td>
<td>3 1/4</td>
</tr>
<tr>
<td>KE-120 Body with pilot cap (125 volts)</td>
<td>2 1/4</td>
<td>3 1/4</td>
</tr>
<tr>
<td>RF-120 Body with Edison adapter cap (660 watts)</td>
<td>2 1/4</td>
<td>3 1/4</td>
</tr>
</tbody>
</table>

**PLATES FOR “SPARTAN” STANDARD FLUSH RECEPTACLES**

These plates are also listed elsewhere for use in connection with other flush receptacles.

<table>
<thead>
<tr>
<th>No. 429</th>
<th>Inside</th>
<th>Outside</th>
</tr>
</thead>
<tbody>
<tr>
<td>420 Single plate, stamped, 1/16 in., 4 1/2 x 2 1/2</td>
<td>25</td>
<td>*</td>
</tr>
<tr>
<td>545 Single plate, solid, 4 1/2 x 3 1/4</td>
<td>25</td>
<td>*</td>
</tr>
<tr>
<td>520 Two gang plate, solid, 4 1/2 x 4 1/4</td>
<td>10</td>
<td>*</td>
</tr>
<tr>
<td>530 Three gang plate, solid, 4 1/2 x 6 1/4</td>
<td>5</td>
<td>*</td>
</tr>
<tr>
<td>531 Four gang plate, solid, 4 1/2 x 8 1/4</td>
<td>5</td>
<td>*</td>
</tr>
</tbody>
</table>

Receptacles in gangs are spaced 1 1/4 inches on centers. A standard package of plates consists of a sufficient number to accommodate 100 receptacles. For special finishes on plates, see listing elsewhere. For special finishes on brass-covered caps, see listing elsewhere. National Electrical Code Standard.

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Wiring Devices
Systems and Standards

• The Electrical System
  – Standards – but not international
  – They didn’t have a clue what we’d be powering in 2005
  – Ubiquity drives prices down

• The Telephone System
  – Even better standards
  – High reliability
  – Not much has changed – and maybe it never will

• The Internet System
  …
What’s SIP

• IETF RFC 3261
  – Replaces RFC 2543

• “The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying and terminating sessions with one or more participants.”

• Can be used for voice, video, instant messaging, gaming, etc., etc., etc.

• Follows on HTTP
  – Text based messaging
  – URIs – ex: sip:dbaron@MIT.EDU
Where’s SIP

Application
- RTSP
- SIP
- RTP
- DNS(SRV)

Transport
- TCP
- UDP

Network
- IP

Physical/Data Link
- Ethernet
**SIP Components**

- **User Agents (UA)**
  - Clients – Make requests
  - Servers – Receive requests

- **Server types**
  - Redirect Server
  - Proxy Server
  - Registrar Server
  - *Location Server*

- **Gateway**
  - UA connecting to another network – eg. the PSTN

- **B2BUAs**
  - Two UAs that pass SIP messages – and can modify them
SIP Trapezoid

- DNS Server
- Location Server
- Registrar
- Outgoing Proxy
- Incoming Proxy
- Originating User Agent
- Terminating User Agent

- SIP
- RTP
SIP Triangle

- DNS Server
- Location Server
- Registrar
- Incoming Proxy
- Terminating User Agent
- Originating User Agent
SIP Peer to Peer!
SIP Methods

- INVITE Requests a session
- ACK Final response to the INVITE
- OPTIONS Ask for server capabilities
- CANCEL Cancels a pending request
- BYE Terminates a session
- REGISTER Sends user’s address to server
## SIP Responses

<table>
<thead>
<tr>
<th>Code</th>
<th>Description</th>
<th>Response</th>
</tr>
</thead>
<tbody>
<tr>
<td>1XX</td>
<td>Provisional</td>
<td>100 Trying</td>
</tr>
<tr>
<td>2XX</td>
<td>Successful</td>
<td>200 OK</td>
</tr>
<tr>
<td>3XX</td>
<td>Redirection</td>
<td>302 Moved Temporarily</td>
</tr>
<tr>
<td>4XX</td>
<td>Client Error</td>
<td>404 Not Found</td>
</tr>
<tr>
<td>5XX</td>
<td>Server Error</td>
<td>504 Server Time-out</td>
</tr>
<tr>
<td>6XX</td>
<td>Global Failure</td>
<td>603 Decline</td>
</tr>
</tbody>
</table>
SIP Flows - Basic

User A

"Calls"
18.18.2.4

INVITE: sip:18.18.2.4

180 - Ringing

200 - OK

ACK

Rings

Answers

Talking

RTP

Talking

Hangs up

BYE

200 - OK

User B
INVITE sip:e9-airport.mit.edu SIP/2.0
From: "Dennis Baron"<sip:6172531000@mit.edu>;tag=1c41
To: sip:e9-airport.mit.edu
Call-Id: call-1096504121-2@18.10.0.79
Cseq: 1 INVITE
Contact: "Dennis Baron"<sip:6172531000@18.10.0.79>
Content-Type: application/sdp
Content-Length: 304
Accept-Language: en
Allow: INVITE, ACK, CANCEL, BYE, REFER, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE
Supported: sip-cc, sip-cc-01, timer, replaces
User-Agent: Pingtel/2.1.11 (WinNT)
Date: Thu, 30 Sep 2004 00:28:42 GMT
Via: SIP/2.0/UDP 18.10.0.79
Session Description Protocol

• IETF RFC 2327

• “SDP is intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation.”

• SDP includes:
  – The type of media (video, audio, etc.)
  – The transport protocol (RTP/UDP/IP, H.320, etc.)
  – The format of the media (H.261 video, MPEG video, etc.)
  – Information to receive those media (addresses, ports, formats and so on)
v=0
o=Pingtel 5 5 IN IP4 18.10.0.79
s=phone-call
c=IN IP4 18.10.0.79
t=0 0
m=audio 8766 RTP/AVP 96 97 0 8 18 98
a=rtpmap:96 eg711u/8000/1
a=rtpmap:97 eg711a/8000/1
a=rtpmap:0 pcmu/8000/1
a=rtpmap:8 pcma/8000/1
a=rtpmap:18 g729/8000/1
a=fmtp:18 annexb=no
a=rtpmap:98 telephone-event/8000/1
CODECs

• GIPS Enhanced G.711
  – 8kHz sampling rate
  – Voice Activity Detection
  – Variable bit rate

• G.711
  – 8kHz sampling rate
  – 64kbps

• G.729
  – 8kHz sampling rate
  – 8kbps
  – Voice Activity Detection
SIP Flows - Registration

User

REGISTER: sip:dbaron@MIT.EDU

401 - Unauthorized

REGISTER: (add credentials)

200 - OK

Registrar

Location

sip:dbaron@MIT.EDU
Contact 18.18.2.4
SIP REGISTER

REGISTER sip:mit.edu SIP/2.0
From: "Dennis Baron"<sip:6172531000@mit.edu>;tag=4561c4561
To: "Dennis Baron"<sip:6172531000@mit.edu>;tag=324591026
Call-Id: 9ce902bd23b070ae0108b225b94ac7fa
Cseq: 5 REGISTER
Contact: "Dennis Baron"<sip:6172531000@18.10.0.79;LINEID=05523f7a97b54dfa3f0c0e3746d73a2
Expires: 3600
Date: Thu, 30 Sep 2004 00:46:53 GMT
Accept-Language: en
Supported: sip-cc, sip-cc-01, timer, replaces
User-Agent: Pingtel/2.1.11 (WinNT)
Content-Length: 0
Via: SIP/2.0/UDP 18.10.0.79
SIP REGISTER – 401 Response

SIP/2.0 401 Unauthorized
From: "Dennis Baron"<sip:6172531000@mit.edu>;tag=4561c4561
To: "Dennis Baron"<sip:6172531000@mit.edu>;tag=324591026
Call-Id: 9ce902bd23b070ae0108b225b94ac7fa
Cseq: 5 REGISTER
Via: SIP/2.0/UDP 18.10.0.79
Www-Authenticate: Digest realm="mit.edu",
    nonce="f83234924b8ae841b9b0ae8a92dcf0b71096505216", opaque="reg:change4"
Date: Thu, 30 Sep 2004 00:46:56 GMT
Allow: INVITE, ACK, CANCEL, BYE, REFER, OPTIONS, REGISTER, NOTIFY, SUBSCRIBE
User-Agent: Pingtel/2.2.0 (Linux)
Accept-Language: en
Supported: sip-cc-01, timer
Content-Length: 0
**SIP REGISTER with Credentials**

```
REGISTER sip:mit.edu SIP/2.0
From: "Dennis Baron"<sip:6172531000@mit.edu>;tag=4561c4561
To: "Dennis Baron"<sip:6172531000@mit.edu>;tag=324591026
Call-Id: 9ce902bd23b070ae0108b225b94ac7fa
Cseq: 6 REGISTER
Contact: "Dennis Baron"<sip:61725231000@18.10.0.79;LINEID=05523f7a97b54dfa3f0c0e3746d73a
Expires: 3600
Date: Thu, 30 Sep 2004 00:46:53 GMT
Accept-Language: en
Supported: sip-cc, sip-cc-01, timer, replaces
User-Agent: Pingtel/2.1.11 (WinNT)
Content-Length: 0
Authorization: DIGEST USERNAME="6172531000@mit.edu", REALM="mit.edu",
    NONCE="f83234924b8ae841b9b0ae8a92dcf0b71096505216", URI="sip:mit.edu",
    RESPONSE="ae064221a50668eaad1ff2741fa8df7d", OPAQUE="reg:change4"
Via: SIP/2.0/UDP 18.10.0.79
```
SIP Flows – Via Proxy

INVITE: sip:dbaron@MIT.EDU
INVITE: sip:dbaron@18.18.2.4
100 - Trying
180 - Ringing
200 - OK
180 - Ringing
200 - OK
ACK
ACK
RTP
Hangs up
“Calls” dbaron
@MIT.EDU
Proxy

User A
User B

Rings
Answers
Talking
Talking

Massachusetts Institute of Technology
SIP Flows – Via Gateway

INVITE: sip:joe@MIT.EDU

INVITE: sip:38400@18.162.0.25

100 - Trying

180 - Ringing

200 - OK

ACK

ACK

RTP

Rings

Answers

Talking

Hangs up

BYE

BYE

200 - OK
SIP INVITE with Record-Route

INVITE sip:37669@18.162.0.25 SIP/2.0
Record-Route: <sip:18.7.21.118:5080;lr;a;t=2c41;s=b07e28aa8f94660e8545313a44b9ed50>
From: "Dennis Baron"<sip:6172531000@mit.edu>;tag=2c41
To: sip:37669@mit.edu
Call-Id: call-1096505069-3@18.10.0.79
Cseq: 1 INVITE
Contact: "Dennis Baron"<sip:6172531000@18.10.0.79>
Content-Type: application/sdp
Content-Length: 304
Accept-Language: en
Allow: INVITE, ACK, CANCEL, BYE, REFER, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE
Supported: sip-cc, sip-cc-01, timer, replaces
User-Agent: Pingtel/2.1.11 (WinNT)
Date: Thu, 30 Sep 2004 00:44:30 GMT
Via: SIP/2.0/UDP 18.7.21.118:5080;branch=z9hG4bK2cf12c563cec06fd1849ff799d069cc0
Via: SIP/2.0/UDP 18.7.21.118;branch=z9hG4bKd26e44dfdc2567170d9d32a143a7f4d8
Via: SIP/2.0/UDP 18.10.0.79
Max-Forwards: 17
SIP SUBSCRIBE

SUBSCRIBE sip:6172531000@mit.edu SIP/2.0
From: "Dennis Baron"<sip:6172531000@mit.edu>;tag=11005c11005
To: "Dennis Baron"<sip:6172531000@mit.edu>;tag=765268780
Call-Id: 9c0a1ef37f461a8feb7b80fe84855a4f
Cseq: 1451 SUBSCRIBE
Contact: sip:6172531000@18.10.0.79
Event: message-summary
Accept: application/simple-message-summary
Expires: 3600
Date: Wed, 05 Jan 2005 02:57:34 GMT
Accept-Language: en
Supported: sip-cc, sip-cc-01, timer, replaces
User-Agent: Pingtel/2.1.11 (VxWorks)
Content-Length: 0
Via: SIP/2.0/UDP 18.10.0.79
SIP NOTIFY

NOTIFY sip:6172531000@18.142.4.231 SIP/2.0
Content-Type: application/simple-message-summary
Content-Length: 47
Event: message-summary
From: "Dennis Baron"<sip:6172531000@mit.edu>;tag=11005c11005
To: "Dennis Baron"<sip:6172531000@mit.edu>;tag=765268780
Call-Id: 9c0a1ef37f461a8feb7b80fe84855a4f
Cseq: 2944 NOTIFY
Contact: sip:18.7.21.118:5110
Date: Wed, 05 Jan 2005 02:57:35 GMT
Max-Forwards: 20
User-Agent: Pingtel/2.2.0 (Linux)
Accept-Language: en
Supported: sip-cc-01, timer
Via: SIP/2.0/UDP 18.7.21.118:5110;branch=z9hG4bK6d9d30fb13e4c32dc6621d480c4882ca

Messages-Waiting: no
Voicemail: 0/70 (0/0)
SIP Standards

Just a sampling of IETF standards work…

IETF RFCs  [http://ietf.org/rfc.html](http://ietf.org/rfc.html)

- RFC3261  Core SIP specification – obsoletes RFC2543
- RFC2327  SDP – Session Description Protocol
- RFC1889  RTP - Real-time Transport Protocol
- RFC2326  RTSP - Real-Time Streaming Protocol
- RFC3262  SIP PRACK method – reliability for 1XX messages
- RFC3263  Locating SIP servers – SRV and NAPTR
- RFC3264  Offer/answer model for SDP use with SIP
SIP Standards (cont.)

- RFC3265 SIP event notification – SUBSCRIBE and NOTIFY
- RFC3266 IPv6 support in SDP
- RFC3311 SIP UPDATE method – eg. changing media
- RFC3325 Asserted identity in trusted networks
- RFC3361 Locating outbound SIP proxy with DHCP
- RFC3428 SIP extensions for Instant Messaging
- RFC3515 SIP REFER method – eg. call transfer
SIP – The Good and Not-so-good

• The Good
  – An open standard
  – End-to-end protocol allows features in the clients
  – Can serve many functions
  – Many implementations

• The Not-so-good
  – Maybe too many standards (RFCs)
  – Not enough conventions for how to build services
  – End-to-end means harder to add features in the “network”
Questions?