SIP Basics

Internet2 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
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

SDP
- codecs

TCP
- RTP

UDP
- DNS(SRV)
SIP Components

• User Agents
  – Clients – Make and receive “calls”
  – Servers – Accept requests

• Server types
  – Redirect Server
  – Proxy Server
  – Registrar Server
  – Location Server

• Gateways
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

• 1XX Provisional 100 Trying
• 2XX Successful 200 OK
• 3XX Redirection 302 Moved Temporarily
• 4XX Client Error 404 Not Found
• 5XX Server Error 504 Server Time-out
• 6XX Global Failure 603 Decline
SIP Flows - Basic

INVITE: sip:18.18.2.4

180 - Ringing

200 - OK

ACK

RTP

Hangs up

"Calls" 18.18.2.4

User A

User B

200 - OK

Talking

BYE

Answers

Talking

200 - OK

180 - Ringing

INVITE: sip:18.18.2.4

Rings

"Calls" 18.18.2.4

User A

User B

200 - OK
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)
SDP

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 B

Registrar

Location

REGISTER: sip:dbaron@MIT.EDU

401 - Unauthorized

REGISTER: (add credentials)

200 - OK

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=05523f7a97b54dfa3f0c0e3746d73a24f
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="ae064221a50668eada1ff2741fa8df7d", 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
ACK

Rings
Answers
Talking
Hangs up

User A
Proxy
User B

RTP

200 - OK
ACK

BYE

200 - OK
ACK

MIT.EDU
MIT.EDU

Talks
Talks
SIP Flows – Via Gateway

INVITE: sip:joe@MIT.EDU
100 - Trying
180 - Ringing
200 - OK
ACK

INVITE: sip:38400@18.162.0.25
100 - Trying
180 - Ringing
200 - OK
ACK

RTP

BYE
200 - OK
200 - OK

Rings
Answers
Talking
Talking

User A
Proxy
Gateway
30161

"Calls" jis
@MIT.EDU

200 - OK

Talking
Hangs up
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 Standards

Just a sampling of IETF standards work…

IETF RFCs  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 authenticated identity management -
Questions?