



Massachusetts Institute of Technology

# SIP Basics

Internet2 VoIP Workshop

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# Outline

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



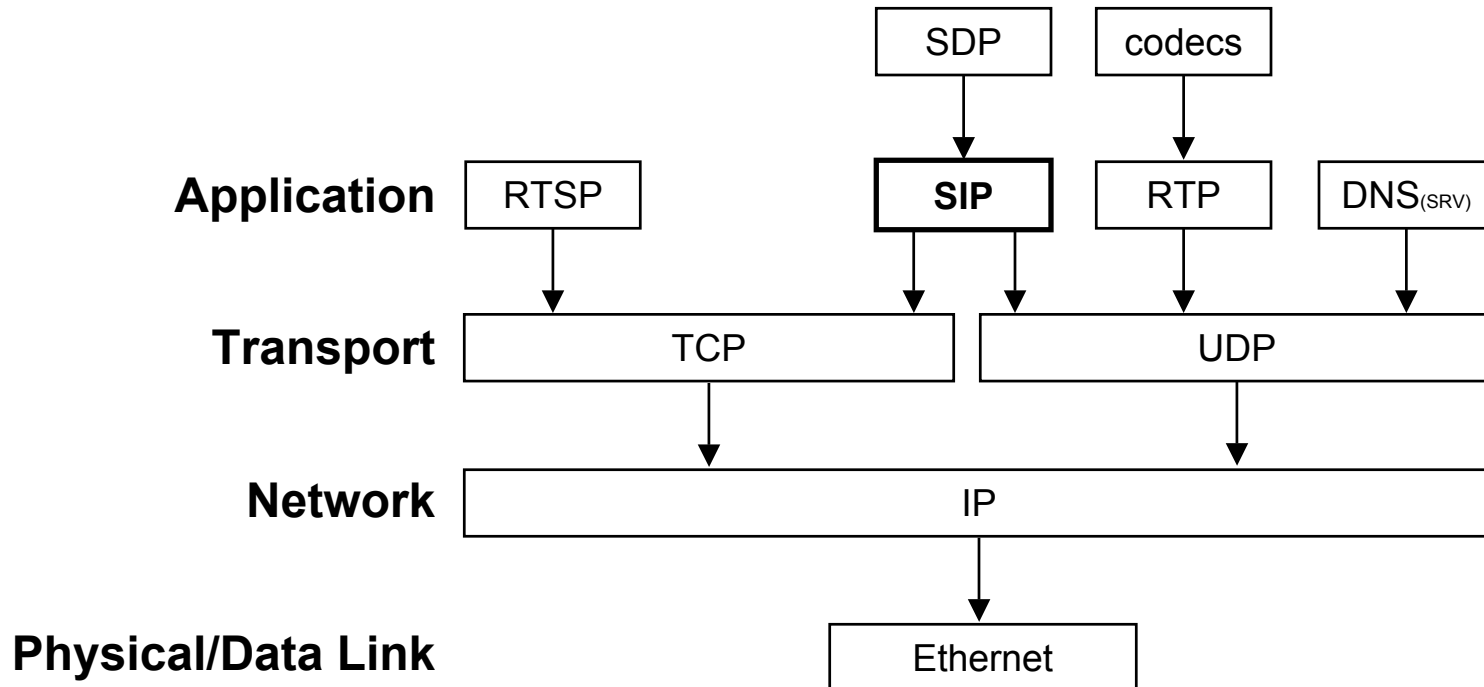
# What's SIP

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- 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



# SIP Components

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- User Agents
  - Clients – Make and receive “calls”
  - Servers – Accept requests
  
- Server types
  - Redirect Server
  - Proxy Server
  - Registrar Server
  - Location Server
  
- Gateways

# SIP Methods

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- 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

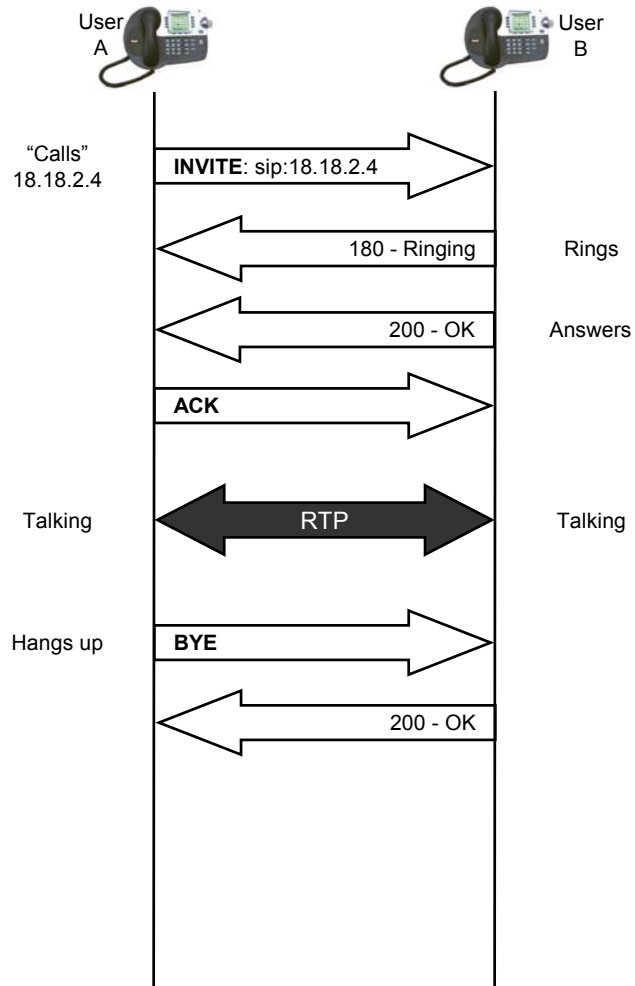


# SIP Responses

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- 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



# SIP INVITE

```
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

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- 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
```

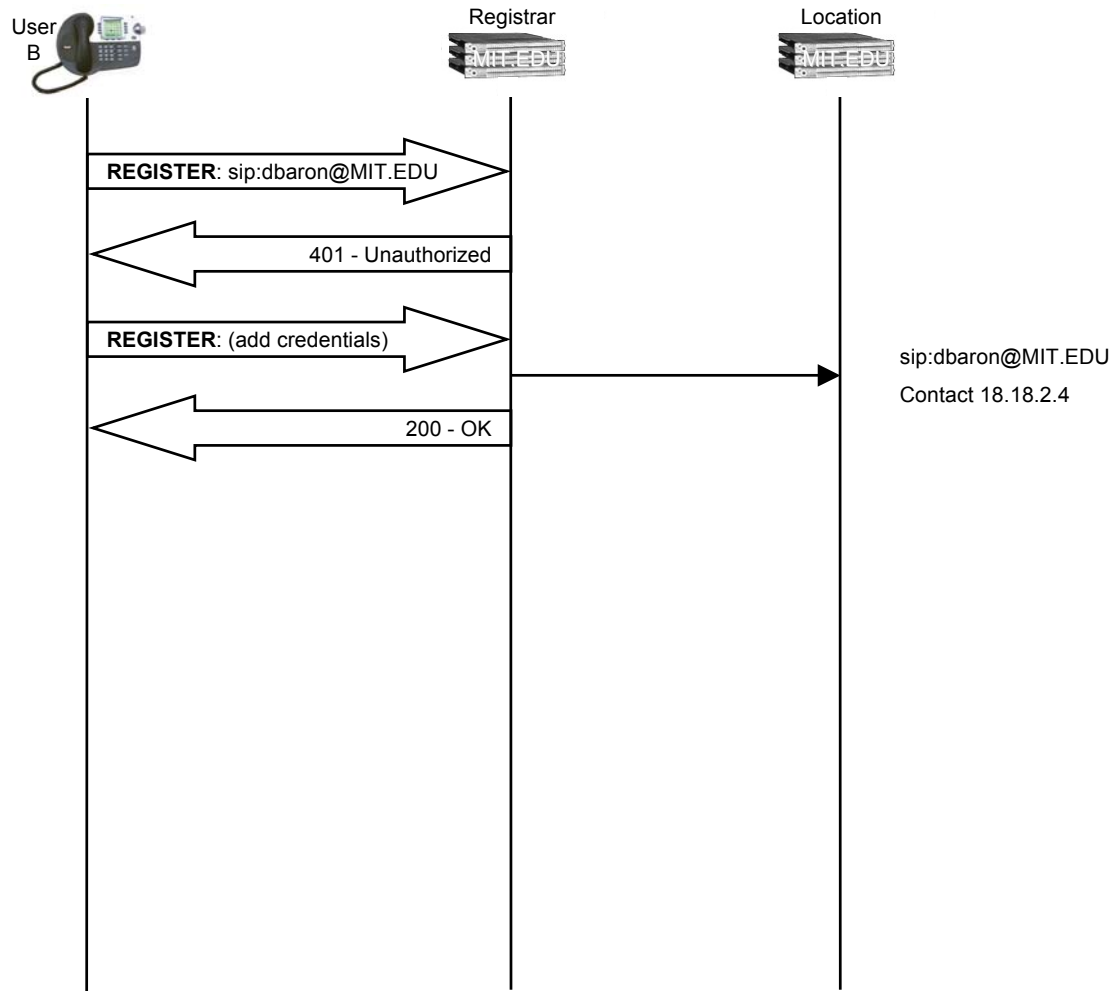


# CODECs

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- 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



# 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=05523f7a97b54dfa3f0c
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, SUBSC

User-Agent: Pingtel/2.2.0 (Linux)

Accept-Language: en

Supported: sip-cc-01, timer

Content-Length: 0

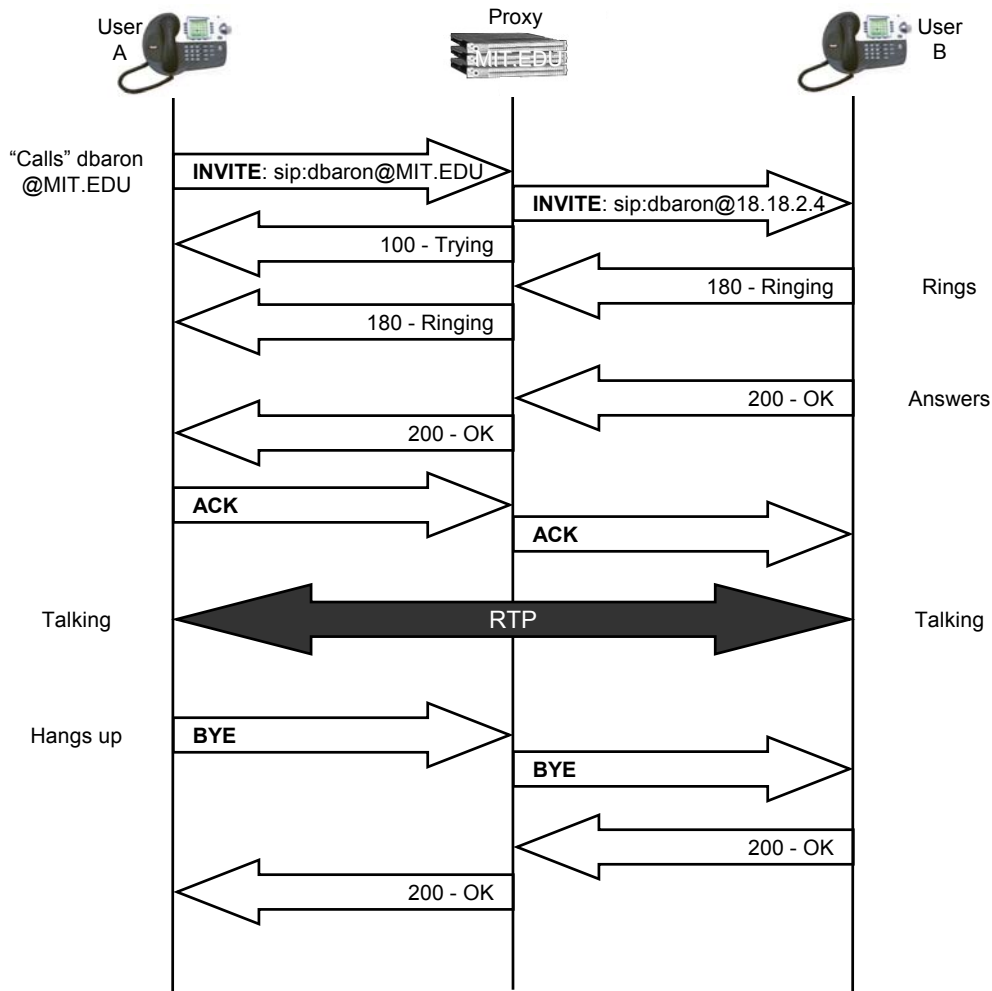


# SIP REGISTER with Credentials

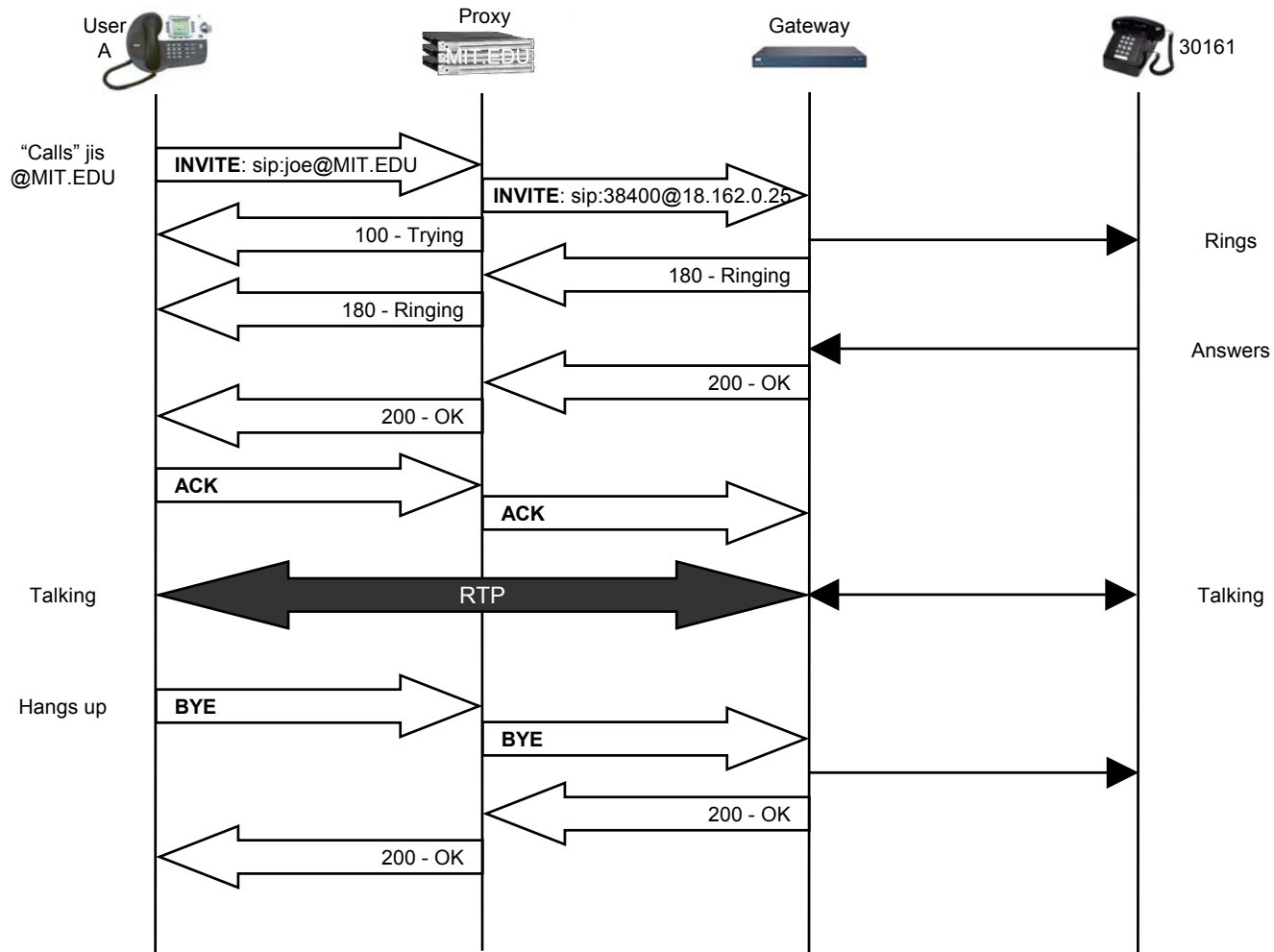
```
REGISTER sip:mit.edu SIP/2.0
From: "Dennis Baron"<sip:6172531000@mit.edu>;tag=4561c4561
To: "Dennis Baron"<sip:6172531000@mit.edu>;tag=324591026
Call-Id: 9ce902bd23b070ae0108b225b94ac7fa
Cseq: 6 REGISTER
Contact: "Dennis Baron"<sip:61725231000@18.10.0.79;LINEID=05523f7a97b54dfa3f0
Expires: 3600
Date: 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



# SIP Flows – Via Gateway



# 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

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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.)

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- 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
- SIMPLE IM/Presence - <http://ietf.org/ids.by.wg/simple.html>
- SIP authenticated identity management -  
<http://www.ietf.org/internet-drafts/draft-ietf-sip-identity-02.txt>



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# Questions?

