Massachusetts Institute of Technology

SIP Basics

CSG VoIP Workshop

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January 5, 2005

Outline

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

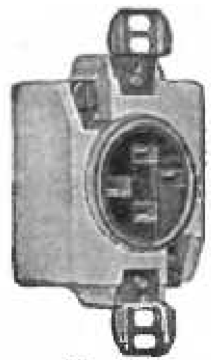
But First…

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

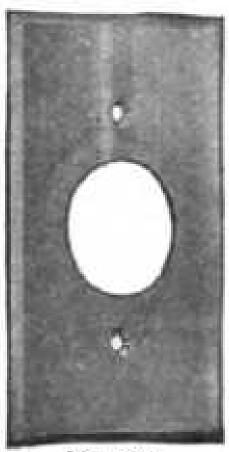
The Electrical System

and predicting the future.

Western Electric Catalog c.1916



No. 120



No. 429

Western-Electric
COMPANY

117	‡Device, complete, without cap	...	4½ x 2¾	3 [illegible]	1	10	..	$2.88
KA-117	‡Body with Standard cap......	...	4½ x 2¾	3 [illegible]	1	10	..	3.10
KB-117	‡Body with brass covered cap..	...	4½ x 2¾	3 [illegible]	1	10	..	3.24
KC-117	‡Body with finger grip cap.....	...	4½ x 2¾	3 [illegible]	1	10	..	3.39
KD-117	‡Body with elongated cap......	...	4½ x 2¾	3 [illegible]	1	10	..	3.24
KE-117	‡Body with pilot cap (125 volts).	...	4½ x 2¾	3 [illegible]	1	10	..	4.25
KF-117	‡Body with Edison adapter cap (660 watts)................	...	4½ x 2¾	3 [illegible]	1	10	..	3.17

"SPARTAN" FLUSH HEATER CONTROL COMBINATION

407 ‡This device is listed elsewhere.

"SPARTAN" STANDARD FLUSH RECEPTACLES—10 AMPERES, 250 VOLTS

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

			Inside	Outside				
120	‡Flush receptacle body........	...	2 3/16	3 9/32	10	50	28	$0.87
KA-120	‡Body with Standard cap......	...	2 3/16	3 9/32	10	50	28	1.08
KB-120	‡Body with brass covered cap..	...	2 3/16	3 9/32	10	50	28	1.23
KC-120	‡Body with finger grip cap.....	...	2 3/16	3 9/32	10	50	28	1.37
KD-120	‡Body with elongated cap......	...	2 3/16	3 9/32	10	50	30	1.23
KE-120	‡Body with pilot cap (125 volts).	...	2 3/16	3 9/32	10	30	..	2.24
KF-120	‡Body with Edison adapter cap (660 watts)................	...	2 3/16	3 9/32	10	30	..	1.16

PLATES FOR "SPARTAN" STANDARD FLUSH RECEPTACLES

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

429	‡Single plate, stamped, 1/16 in., 4½ x 2¾..................	...	...	...	25	*	..	$0.51
545	‡Single plate, solid, 4½ x 3¾..	...	...	...	25	*	..	.72
529	‡Two gang plate, solid, 4½ x 4 9/16	...	...	...	10	*	..	1.44
530	‡Three gang plate, solid, 4½ x 6 3/8	...	...	...	5	*	..	2.16
531	‡Four gang plate, solid, 4½ x 8 3/16	...	...	...	5	*	..	2.88

Receptacles in gangs are spaced 1 13/16 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.

32-4 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

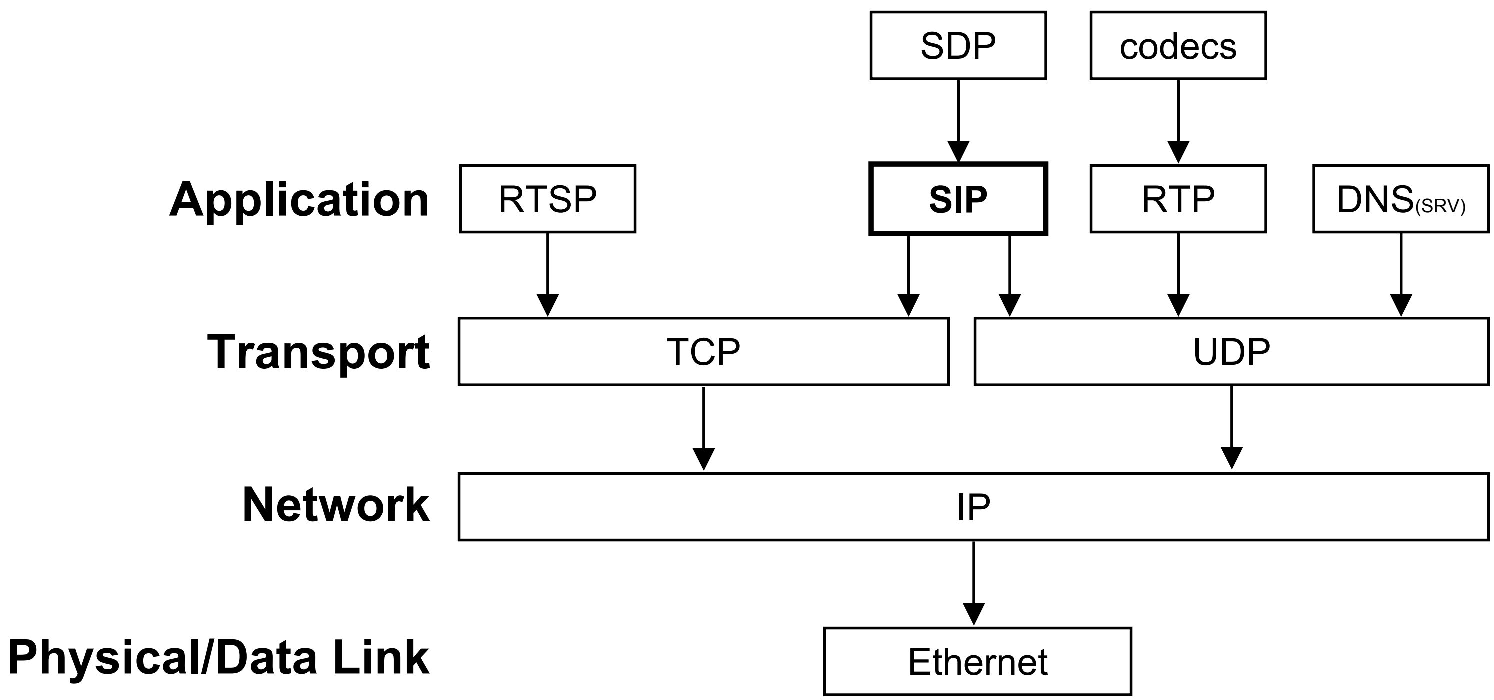
- The Internet System

  …

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

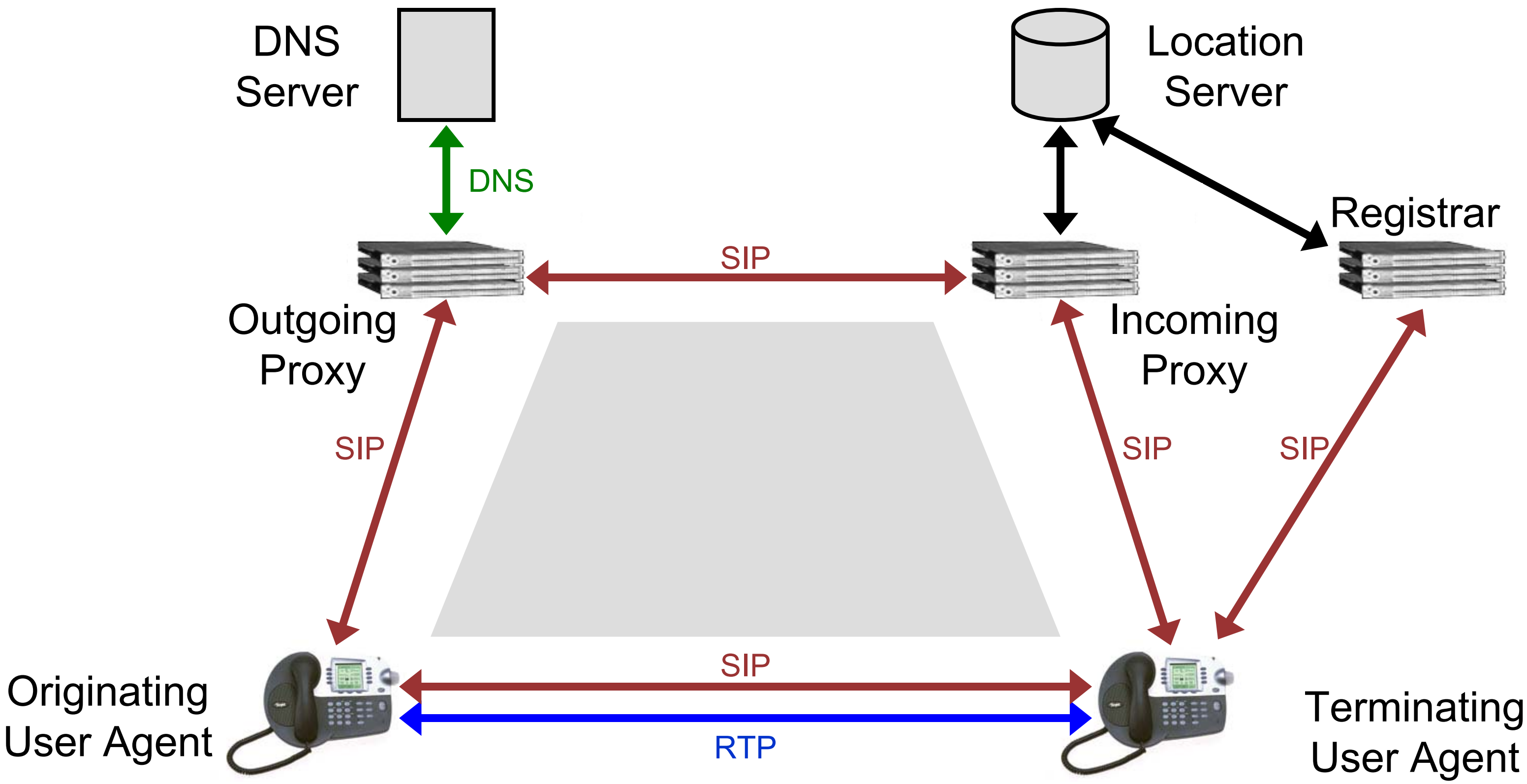
Where's SIP


SDP
codecs
Application
RTSP
SIP
RTP
DNS(SRV)
Transport
TCP
UDP
Network
IP
Physical/Data Link
Ethernet

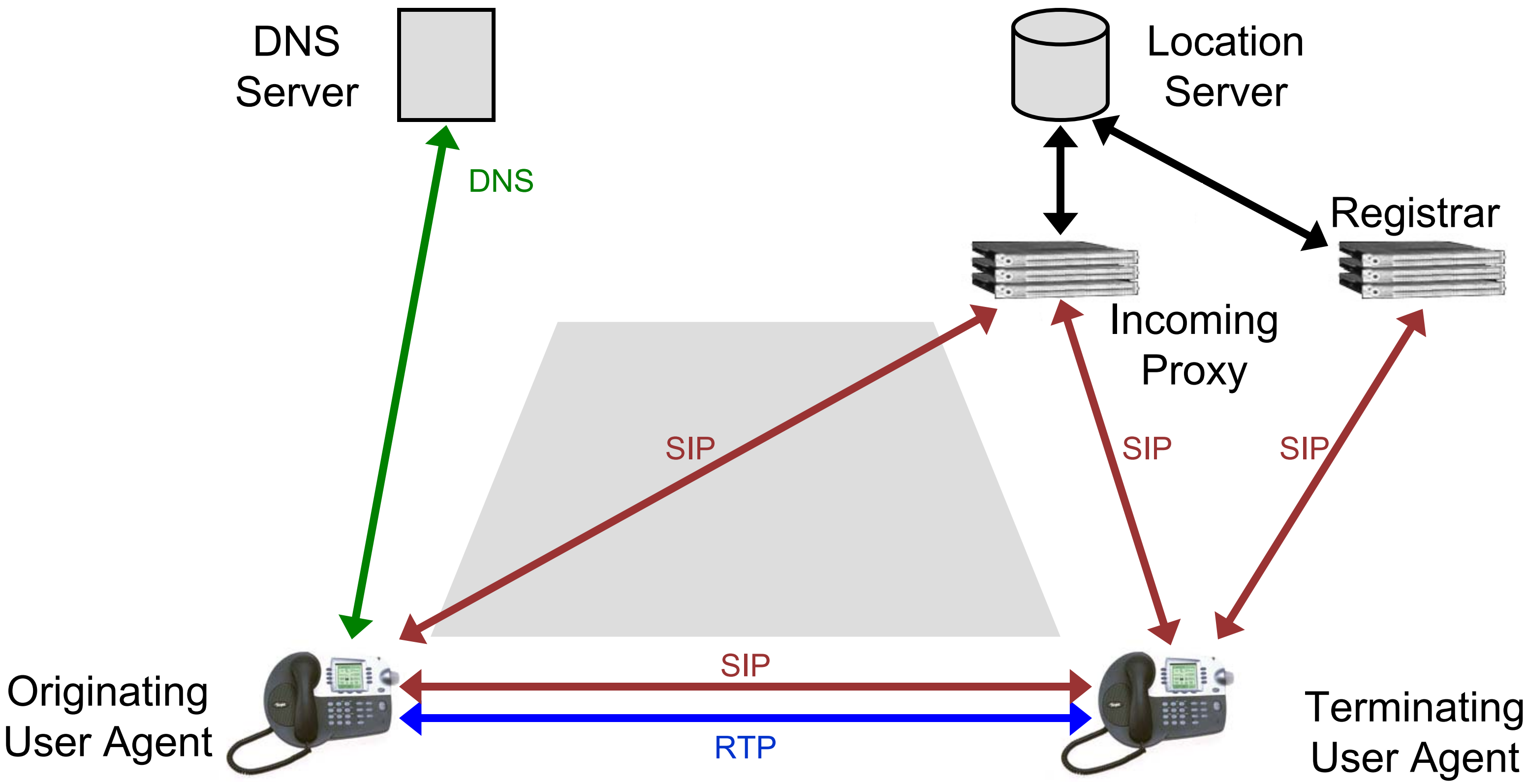
SIP Components

- 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

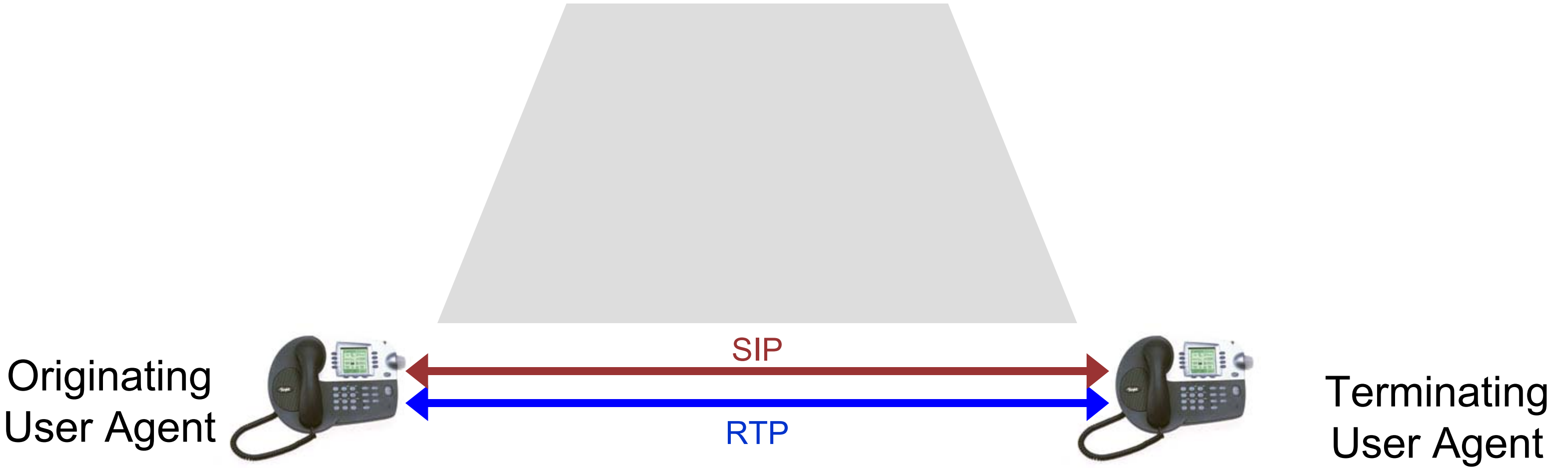
SIP Trapezoid



SIP Triangle



SIP Peer to Peer!



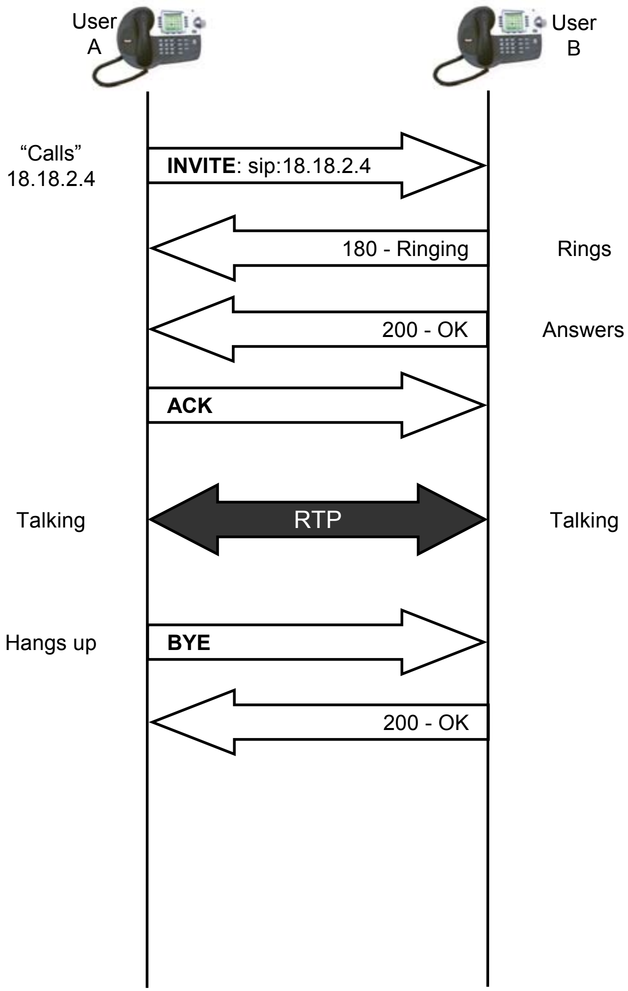
SIP Methods

- 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

• 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

- 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)

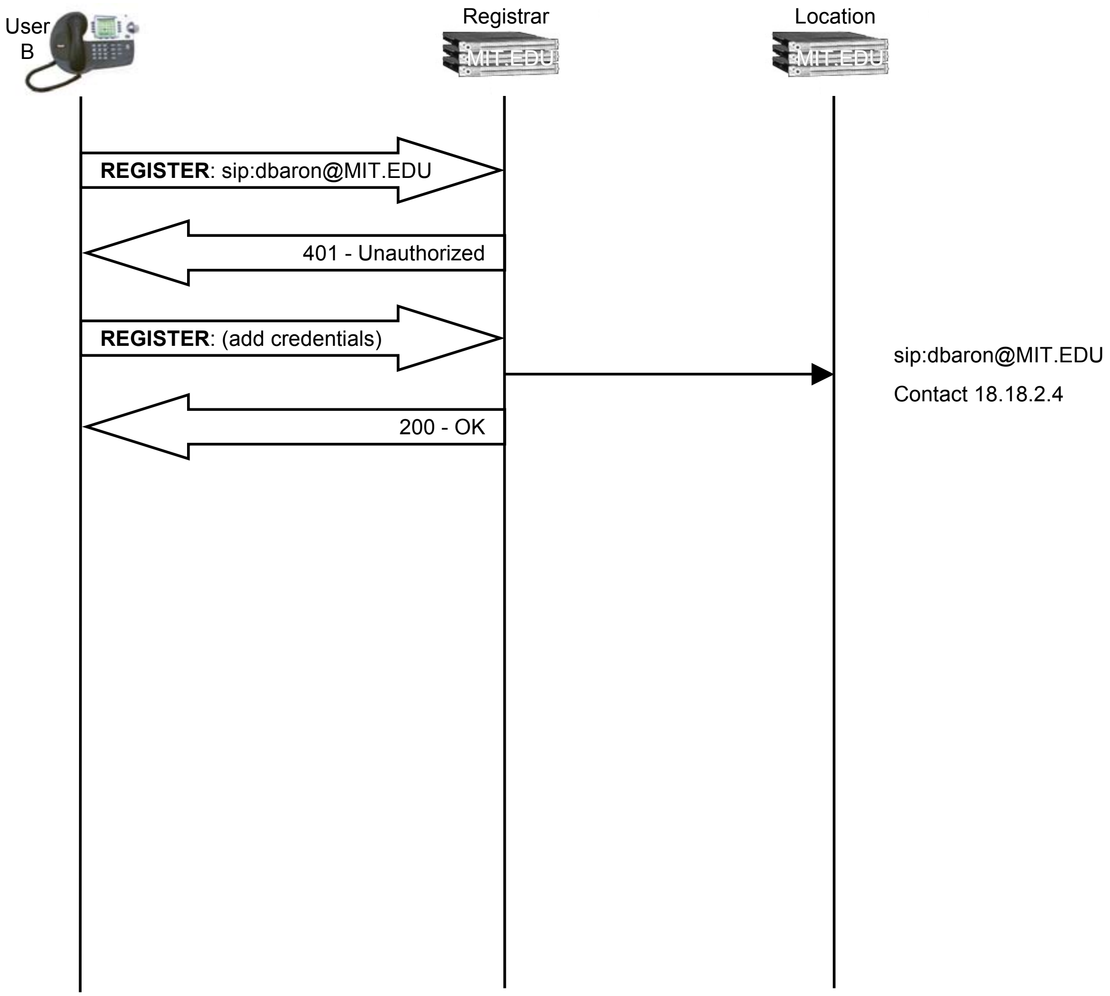
SDP

```
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



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, SUBS

User-Agent: Pingtel/2.2.0 (Linux)

Accept-Language: en

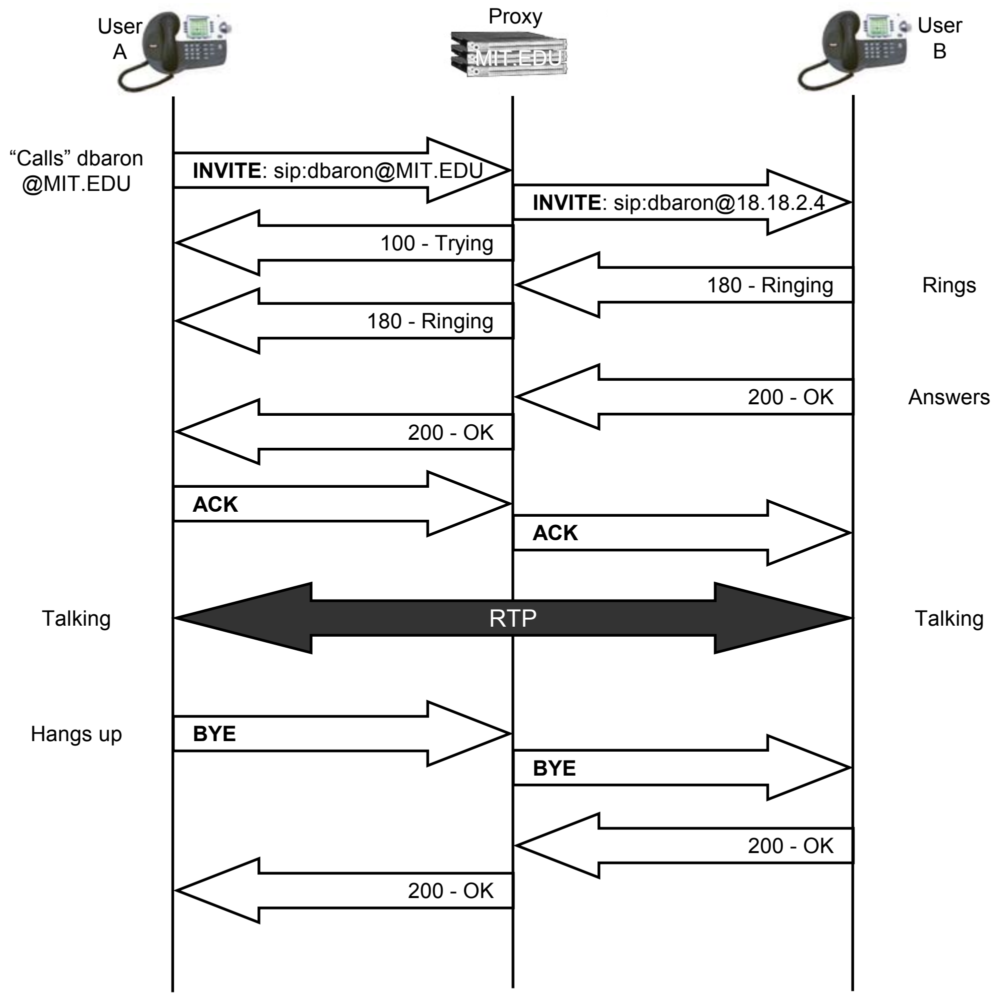
Supported: sip-cc-01, timer

Content-Length: 0

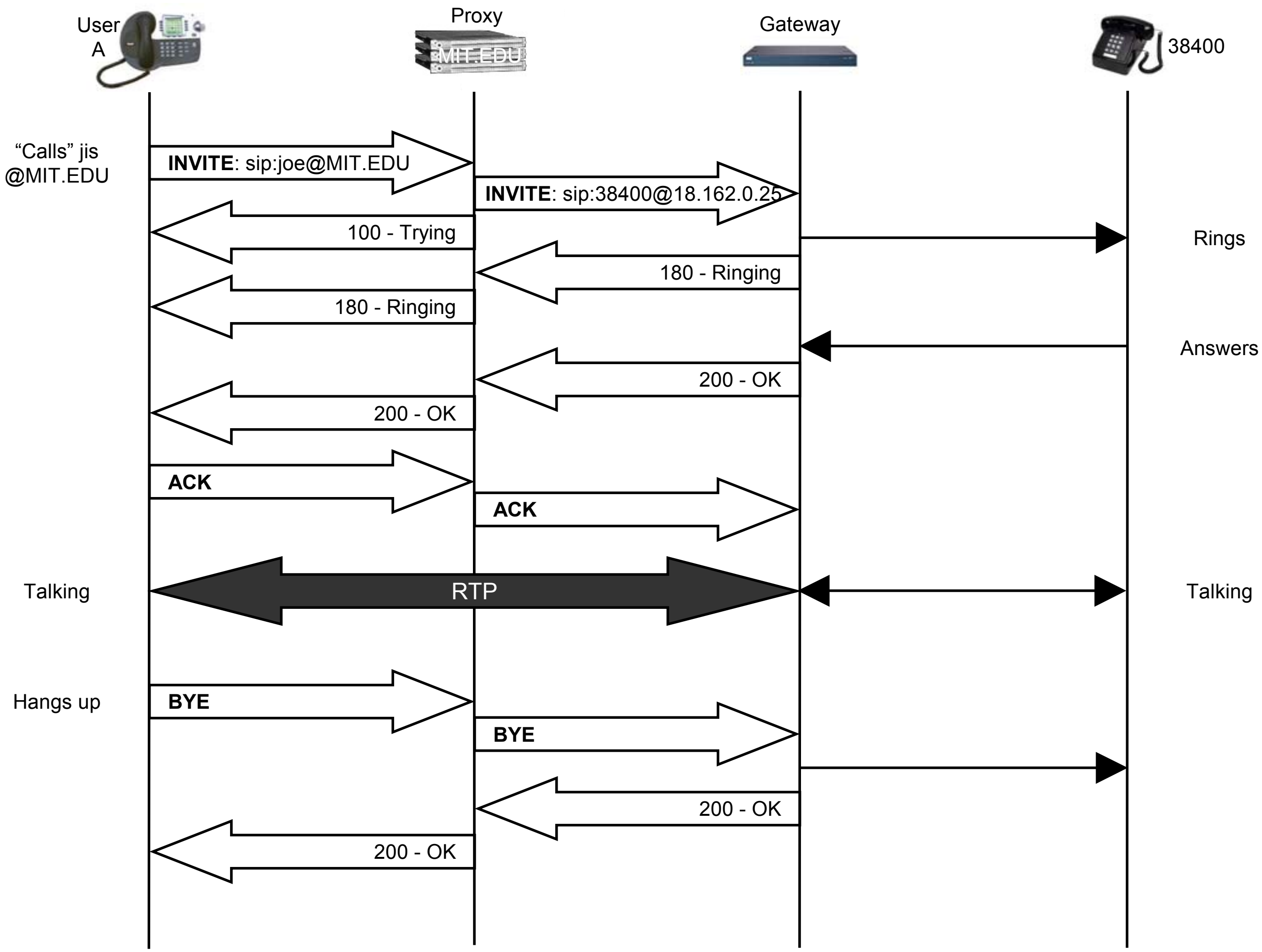
SIP REGISTER with Credentials

```
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 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

- 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
- 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

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?