



## SESSION INITIATION PROTOCOL “A truly disruptive technology”

H. Charles Baker

### Outline of this Tutorial

- Introduction – SIP: What and Why?
- Characterizing the Internet
- Historic Problems with VoIP
- What SIP is and what it does
- SIP and Related Protocols, and their relationship to TCP/IP and UDP/IP
- Why SIP eliminates past performance problems of VoIP
- How companies can evolve to SIP
- How you can try SIP at home
- Getting more info on SIP

### Definition of SIP

Session Initiation Protocol is an umbrella signaling protocol that is used to establish and terminate voice and multimedia communications across the Internet.

- Protocol – A set of rules governing a time sequence of events
- Signaling – Control of communications
- Umbrella – Over other protocols

SIP is emerging as the protocol of choice for setting up conferencing, telephony, multimedia and all other types of communication sessions on the Internet. SIP may also be used for new types of communications, such as instant messaging, presence and application level mobility across various networks, including wireless, and across user devices.

Work on SIP is accomplished primarily in the IETF SIP working group. SIP has manifold interactions with other areas, such as the next generation wireless internetworks, QoS, payments and security.

### Some SIP Milestones

- SIP has been selected by Microsoft as its protocol of choice for development of VoIP. All Windows XP shipped after Sept. 30, 2002 is SIP enabled for Windows Messenger.
- 3GPP has selected SIP for development of 3G Wireless and has released specifications to Motorola and Nokia.
- Verizon recently announced a multi-year contract to Nortel Networks for development of SIP-compatible CO and networking equipment

### More SIP Milestones

- VoIP using SIP was recently featured on ABC's World News Tonight and other news programs
- Some 400 companies are developing SIP-based products.
- Numerous Fortune 1000 companies are using SIP to reduce communications costs for intra-company calls.
- Over 150,000 residential customers are using SIP for all long-distance calls.

## Because of its distributed intelligence, SIP

- provides every desirable service available on PSTN and PBX's, plus a lot more
- is more flexible than PSTN
- allows rapid development and deployment of new services, including self-defined services through the Call Processing Language (CPL)
- does not have the drawbacks of centralized intelligence on the PSTN
- is "the third great protocol of the Internet, after TCP/IP and HTTP" (Vinton Cerf)

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## Value Proposition of IP Communication Services

- Multimedia: Text, voice, video, data
- Event based communications
- Multiple conferencing models and media
- Secure communications
- User preferences and control for all of the above
- Integration with the Web
- Gateways to PSTN, H.323, paging networks, etc
- 100% open standards, vendor interoperable
- Service development is easy and fast
- Bottom line: Lowest overall cost and highest functionality combined

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## VoIP is better than VoPSTN!

- It's time to lay to rest the notion that ISP's can't deliver telephone-company-level reliability\*
- The Internet proved more reliable than PSTN on 9/11/01 and on 8/15/03
- Most major IP providers exceed the vaunted 5-9's standard for uptime during normal conditions\*
- Better voice quality than PSTN
- Voice packet loss ( $\approx 1/50$  second) of U.S. ISPs typically averages less than 0.01%\*

\*[www.nwfusion.com/research/2002/1216isptest.html](http://www.nwfusion.com/research/2002/1216isptest.html)

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## Quantifying Today's Internet

- "ping" provides useful transit delay information
- "traceroute" provides useful hop information
- Hosts I checked from SMU Engineering:

Host	# R/T Routers	Avg. R/T Time (ms)
mail.smu.edu	10	2
kolind.dk	48	157
online.ru	38	167
africaonline.co.ug	60	671 (by satellite)

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## Conclusions about Today's Internet

- Very Fast
- Short Transit Delay
- High Capacity
- Robust
- Mostly U.S.
- Internet usage in other countries is growing rapidly

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## Trend in Network Backbones

- Telecom trunks that were formerly part of PSTN are being cutover to Internet trunks
- This trend started in the late 90's in trans-Pacific trunks and is continuing throughout PSTN
- This trend is now accelerating in enterprise networks; i.e., from private lines to Intranet backbones (more on this later)

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## Historic Problems with VoIP

- How many tried VoIP in the mid-to-late 90's?
- Problems:
  1. Long delay
  2. Poor quality audio
- Reasons:
  1. Most delay (500 ms in each direction) was in Windows (pre-XP)
  2. Voice was over-compressed to fit into the 28,800 b/s data circuits

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## Solving the Historic Problems with VoIP

- Windows XP provides only 70 ms delay end-to-end instead of 500 ms (IF you use a soft client)
- Using hard client (analog terminal adaptors -- ATAs), the delay is shorter ( $\approx 40$  ms). This is the packet (datagram) buffer delay
- Use of DSL or cable modem access to the Internet allows a full 64-kb/s  $\mu$ -Law or A-Law encoding

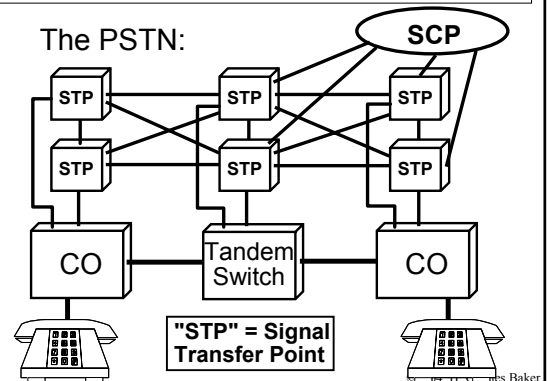
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## What SIP is and What It Does

- SIP discovers remote users and establishes interactive communications sessions
- SIP commands or "requests" (called "Methods") carry with them the necessary underlying protocols to establish desired capabilities in each situation
- After the session is established, and all of the desired parameters negotiated, SIP drops out of the picture until it is desired to modify or terminate the session

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## SIP relates to VoIP as SS7 relates to PSTN



## The SIP Protocol

- SIP is a text-based protocol based on HTTP
- It also uses the same response code classes as HTTP:
  - 1xx – Progress signals and information
  - 2xx – Success
  - 3xx – Redirection
  - 4xx – Client error
  - 5xx – Server error
  - 6xx – Global failure

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## The SIP Methods (requests) – Part 1

- INVITE – Session Setup
- ACK – Acknowledgment of final response to INVITE
- BYE – Session termination
- CANCEL – Cancels pending session
- REGISTER – Registration of a user's URL
- OPTIONS – Query id options and capabilities
- INFO – Mid-call signaling transport
- PRACK – Provisional response acknowledgment

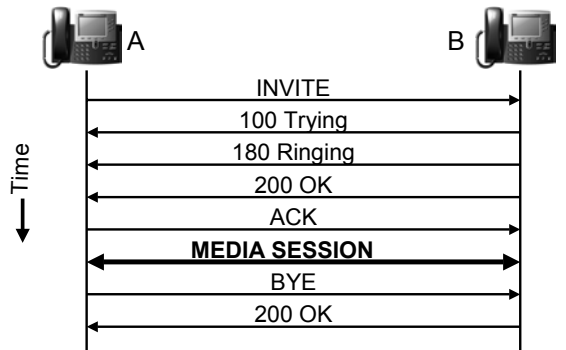
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## SIP Methods – Part 2

- COMET – Notification of preconditions met
- REFER – Transfer user to a URL
- SUBSCRIBE – Request notification of an event
- UNSUBSCRIBE – Cancel SUBSCRIBE
- NOTIFY – Transport of a subscribed event notification
- MESSAGE – Transport of the body of an instant message

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## Protocol for Simple Phone Call (A to B)



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## Each SIP Method has specified parameters, called “Headers”

- Example: INVITE Headers
  - INVITE sip: UserB@there.com SIP/2.0
  - Via: 4.3.2.1:5060
  - To: UserB@there.com
  - From: User A <sip:UserA@here.com>
  - Call-ID:4598998103413@4.3.2.1
  - Cseq: 1 INVITE
  - Contact: <sip:UserA@4.3.2.1
  - Content-Length: 126

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## Session Description Protocol (SDP) provides for Media Negotiation

- SDP follows immediately after the last INVITE header:  
c=IN IP4 128.2.3.1 (Connection=Internet, IPv4, address)  
m=video 4004 RTP/AVP 14 26 (video port #, profiles)  
a=rtpmap:14 MPA/90000 (attributes of video codec & sampling rate)  
a=rtpmap:26 JBEG/90000 (attributes of alt. codec)  
m=audio 4006 RTP/AVP 0 4 (audio port #, profiles)  
a=rtpmap:0 PCMU/8000 (attributes of audio codec & sampling rate)  
a=rtpmap:4 GSM/8000 (attributes of alternate codec)

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## The 200 OK Response declines video, accepts audio, selecting the GSM codec:

- This SDP follows the 200 OK:  
c=IN IP4 16.22.3.1  
m=video 0 RTP/AVP 14 (port “0” declines)  
m=audio 6002 RTP/AVP 4  
a=rtpmap: 4 GSM/8000

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## SIP Uniform Resource Indicators (URIs)

- SIP provides great flexibility of addressing formats
- Each format is called a URI
- Each URI should be able to be converted to a URL
- Example of calling a PSTN Phone:  
sip:+1-972-555-1234@smu.edu;user=phone
- Example of an intranet call:  
sip:123-4567@smu.edu;user=phone;  
context=privnet

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## Other URI Examples

- To call my PC:  
sip:baker@engr.smu.edu
- To establish a secure link:  
sips:baker@engr.smu.edu
- To call a guest laptop logged into a LAN in a classroom:  
sip:guest314@engr.smu.edu

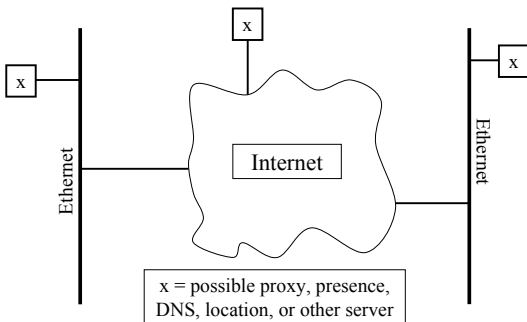
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## SIP Proxy Servers

- Many SIP users register with a proxy server before making calls
- A proxy server can convert URI's to specific IP addresses, sometimes using a Domain Name Server (DNS)
- Proxy servers can also store and impose user preferences for both outgoing and incoming calls
- Proxy Servers can assist presences in providing presence services

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The beautiful thing about the Internet is that you don't care where the servers are



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## Some of the Protocols under SIP (RFC 3261)

- SDP -- Session Description Protocol (RFC 2327)
- RTP – Real-Time Protocol (RFC 1889)
- RSVP – Resource Reservation Protocol (RFC 2205)
- RTSP – Real-time Streaming Protocol (RFC 2326)
- SAP – Session Announcement Protocol (RFC 2974)
- PINT – PSTN & Internet Telephony Services (RFC 2848)
- Spirits – Servers in the PSTN Initiating Requests to Internet Servers (RFC 3136)
- TRIP – Telephone Routing over IP (RFC 3219)

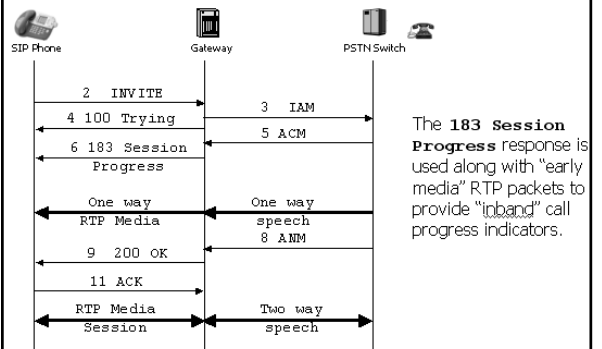
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## PINT Example: IP to PSTN

- PINT requires an extension of the SDP following the INVITE Method:
  - c=TN RFC3261 +1-214-768-3048
  - TN means "Telephone Network"
  - RFC3261 means SIP
- PINT allows converting to FAX context:
  - c=TN RFC3261 +1-214-768-3048
  - m=text 1 fax plain
  - a=fmtp:plain uri:[URL of material to be faxed]

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## A PINT Protocol Example



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## Presence Service is made possible by SIP

- Allows a “Watcher” to find and contact a person who is one the road (the “Presence”) if that person wants to be found.
- Both parties must subscribe to the service before it will work
- Presence service allows the watcher to call the presence as soon as he signs on to the network via any device – examples: phone, PC, PDA, laptop, SIP Phone, etc.

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## Other Services Made Possible by SIP

- Instant Messaging (SIMPLE – SIP for Instant Messaging Protocol Leveraging Extensions)
- Click to dial
- Click to connect
- Re-INVITE (changes parameters of session during session)
- QoS – Quality of Service
- All AIN and modern PBX Features
- It will not replace the undesirable features of PSTN, such as UNP (Universal Numbering Plan) and inability to change the destination of a call

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## A Note About Quality of Service

- In a digital system, QoS refers to nothing more than minimum bit rate
- SIP provides QoS using the PRACK and COMET methods, combined with a QoS attribute under the appropriate media line in the SDP
- Thus the following establishes a minimum level of quality:

```
m=audio 47172 RTP.AVP 0
a=rtpmap:0 PCMU/8000
a=qos:mandatory
```

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## PBX/Centrex-Style SIP Features

Call hold	Consultation hold
Attended call transfer	Attended transfer
Unattended call transfer	3-way conference
Call forwarding/unconditional	Adding a 3 <sup>rd</sup> party
Call Forwarding/busy	3 <sup>rd</sup> party joins
Call forwarding/no answer	Find-me
Single line extension	Call park
Call pickup	Automatic redial
Outgoing call screening	Click to dial
Music on hold	

Ref: <draft-ietf-sipping-service-examples-04.txt>

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## Present Limitation on SIP QoS

- Although a minimum bandwidth can be established at the start of a call, bandwidth theoretically can deteriorate during the progress of a call, due to traffic changes and congestion
- In order to work properly, QoS needs the benefit of IPv6 which contains a built-in RSVP capability

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## Is IP Telephony as Good as PSTN?

- SIP to SIP with 64 kb/s and DSL or cable modem service clearly sounds better than PSTN
- PSTN to PSTN with IP gateways in the middle sounds exactly the same as PSTN
- Dial-access SIP is clearly inferior sound quality, especially for a female speaker, due to over-compressed speech at 8000 b/s or less

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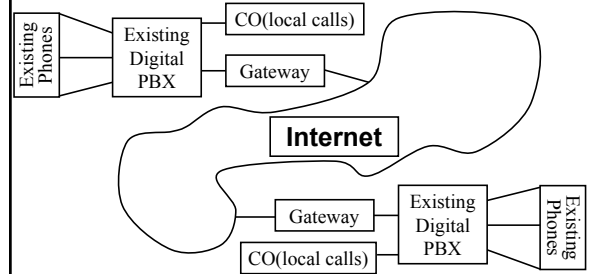
## Voice using SIP will not do:

- Power to telephone from CO
- Precise location info for 911 calls (but static IP phone can be configured with location info) This problem hasn't been solved for cell phones after 20 years. GPS may soon be the answer, both for cell and SIP
- Circuit-like QoS in oversubscribed networks

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## How Companies are Evolving to SIP

- Gateways can be placed between a PBX and the Internet, at low cost per telephone



- Private Internets are secure

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## How You can try SIP (If you have DSL or Cable Modem access to the Internet)

### 1. To try it at no cost:

- Go to XTEN.com, and download the X-Lite soft client
- Use external mike and speakers with your computer to avoid echo to the other end of call
- Subscribe to Free World Dialup at <http://www.pulver.com/fwd>
- Follow instructions on the website

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## How You can try SIP (If you have DSL or Cable Modem access to the Internet)

### 2. To convert to SIP for local and long distance calls:

- Go to <http://www.vonage.com> and subscribe (\$14.99 - \$34.99/month)
- Vonage will provide you with an Analog Terminal Adaptor which will handle two analog lines in your house. Vonage has gateways throughout the world
- Free SIP calls in the U.S. and Canada; small charge to other countries

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## The main SIP Description: RFC 3261

### Other important SIP Method extensions:

SUBSCRIBE/NOTIFY	"SIP Events" RFC 3265 by A. Roach
REFER	"SIP REFER Method" Internet Draft by R. Sparks
MESSAGE	"SIP Extensions for Instant Messaging" Internet Draft by J. Rosenberg <i>et al</i>
PRACK	"Reliability of Provisional Responses in SIP" RFC 3262 by J. Rosenberg & H. Schulzrinne
UPDATE	Early Media and Preconditions: "The SIP UPDATE Method" I-D by J. Rosenberg
INFO	Mid-call signaling information: "The SIP INFO Method" RFC 2976 by S. Donovan

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## IETF Resources on SIP

Finding a draft	<a href="http://search.ietf.org">http://search.ietf.org</a>
Finding an RFC	<a href="http://www.rfc-editor.org/rfcsearch.html">http://www.rfc-editor.org/rfcsearch.html</a>
SIP WG	<a href="http://ietf.org/html.charters/sip-charter.html">http://ietf.org/html.charters/sip-charter.html</a>
SIPPING WG	<a href="http://ietf.org/html.charters/sipping-charter.html">http://ietf.org/html.charters/sipping-charter.html</a>
SIMPLE WG	<a href="http://ietf.org/html.charters/simple-charter.html">http://ietf.org/html.charters/simple-charter.html</a>
XCON WG	<a href="http://ietf.org/html.charters/xcon-charter.html">http://ietf.org/html.charters/xcon-charter.html</a>
MMUSIC WG	<a href="http://ietf.org/html.charters/mmusic-charter.html">http://ietf.org/html.charters/mmusic-charter.html</a>

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## Online Resources fo SIP

- Henning Schulzrinne's SIP Page:
  - <http://www.cs.columbia.edu/sip>
- ietf.org and supplementary SIP and SIPPING archives:
  - <http://www.softarmor.com/sipwg/drafts>
  - <http://www.softarmor.com/sipping/drafts>
- IP Telephony Webpage (includes SIP tutorials)
  - <http://iptel.org>
- SIP Forum:
  - <http://www.sipforum.com>
- SIP Center:
  - <http://www.sipcenter.com>
- SIP Products:
  - <http://www.pulver.com/sip/products.html>

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## More Resources for SIP

- Books
  - Sinnreich & Johnston, Internet Communications Using SIP, Wiley, 2001, ISBN #0-471-41399-2. (This is the textbook used for the SIP Course at SMU.)
  - Johnston, SIP: Understanding the Session Initiation Protocol, Artech House, 2001
  - Camarillo, SIP Demystified, McGraw-Hill, 2001

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## Glossary

- AIN – Advanced Intelligent Network
- AVP – Audio-Video Profile for conferencing
- Bandwidth – In this context, bit rate
- Codec – Device for converting A to D and back
- MMUSIC – Multiparty MULTimedia Sesslon Control
- PDA – Personal Digital Assistant (e.g., Palm)
- Presentity – Server providing presence info
- PSTN – Public Switched Telephone Network (the legacy network)

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## More Glossary

- QoS – Quality of Service
- RFC – Request for Comments
- SIMPLE – SIP for Instant Messaging Protocol Leveraging Extension
- SIPPING – SIP Protocol INstiGation (of applications)
- UNP – Universal Numbering Plan
- URI – Uniform Resource Indicator
- Watcher – Software that requests presence info
- WG – Working Group
- XCon – Centralized Conferencing

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