Introduction to SIP and VoIP

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David Ahlgren

Technology and Business Development

Agenda

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- Hello, Schedule & Hand outs
- SIP Overview, Status & Standards
- SIP Design Concepts and Architectures
- SIP Messaging Details
- VoIP Overview and Architectures
- Applying VoIP
- Network Considerations
- VoIP/QoS Protocols
- Q&A – How much time?

What is SIP ?-

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- A power full new communication method
  - IP network based
  - Enables new forms of communications between people and/or machines
  - Widely adopted and extremely flexible.
- Focused on but not limited to Multimedia
  - Voice, Video etc.
- Written in clear text, human readable form
  - Easy to understand and implement
- The Secret –
  - It is a Control Protocol not a media transport protocol
  - Distributed architecture

How can SIP be used?

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- IP Telephony – Voice & Video
- Instant Messaging
- Music and video on demand
- Interactive gaming
- Calendars, meetings, conf calls
- Full service web Conferencing
- Location based services

SIP Based Products

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- SIP VoIP Phones
  - Pingtel, Cisco, Polycom
- SIP Analog Phone Adaptors - ATA
  - ATT, Vonage, Linksys, SIPphone
- SIP Soft phones
  - Windows -X-Lite, Hotsop
  - Linux -Kphone, PhoneGAIM, LinPhone
- SIP Presence and Messaging services
  - HotSIP, IBM, Sleipner
- SIP Gateways and Softswitches
  - Cisco, Mediatrox,
- SIP Applications Servers
  - HotSIP, Pingtel, Cisco, RadVision, Hughes Networks, Ubiquity,

Who is using SIP?

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ATT	Call Advantage, VoIP, US & Canada, $30/mo
Vonage	VoIP, US & Canada, $25/mo, 500,000 Subscribers!
SIPphone	VoIP US & Canada, $0/mo, $0.02/min for PSTN calls
Microsoft	MSN Chat, Windows XP softphone (350M users in 2005)
IBM	Lotus Instant messaging
HP	HP Open Media Platform
Apple	iChat
Sun	Java API's for component and application development
Cable TV	TW, Instant messaging, Voice and Video, presence
3G cellular	Control protocol for next generation cell phones
Clearwire	(Craig McCaw) VoIP roll out, $100M from Bell Canada
Carriers	Worldcom, Song Networks, Telia, Delta Three, Level3 AT&T, Radiant Telecom and BT

Who Invented SIP?

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- Authors:
  - Henning. Schulzrinne, Columbia U., M. Handley, ACIRI, E. Schooler, Cal Tech, J. Rosenberg, Bell Labs
- Original SIP library.
  - Derived from Columbia's early proxy, registration and redirect servers beginning in 1996
- Developed within the IETF
  - Internet Engineering Task Force
- RFC2543/ RFC3261
  - "SIP: Session Initiation Protocol", March 1999

SIP Standardization Efforts

- One of the most active working groups in the IETF
- Located in “transport” area, but really an application
- 80 active Internet drafts related to SIP
- Typically, 400 attend WG meetings at IETF
- Challenging environment
  - Follows 80-20 rule – 80% of work, 20% of time

Supported by Multiple Standards Groups

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IETF	Internet Engineering Task Force
IETF PINT	PSTN Internet Interfaces WG, Enhance the interface between the PSTN and the Internet
3GPP	3G Preferred Partners, Mobile Specification to support global communication roaming
IPCC	International Packet Communication Consortium, Accelerate VoIP over converged networks – wired, wireless, cable
Packet Cable	DCS - Distributed Call Signaling, Supports call management for packet based voice and multimedia

Related IETF Working Groups

SIMPLE	SIP for Instant Messaging and Presence Language Extensions
MMUSIC	Multiparty Multimedia Session Control
QoS Related	DiffServ, IntServ, RSVP
IPTEL	Internet Telephony
Megaco, SigTran	PSTN legacy
AVT	Audio Video Transport
MIDCOM	Firewall/NAT Traversal

A Remarkably Fast Timeline !

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1996	Academic R&D	First technology concept drafts
1999	Skunk works	RFC 2543 –submitted to the IETF
1999		First SIP bake-off*
2000	Productize	SIP becomes 3GPP signaling protocol
2001	Standardize	RFC 3261 – Official standard (obsoletes 2543)
2002	Mature	World wide bake-off, first time outside U.S.
RFC - Request for Comments, Official method for proposing to the IETF		
* Pillsbury sued Columbia et al on use of the term BakeOff !!		

SIP - A New Generation of Communication Services

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- SIP Extends the ITEF spirit of open standards
- Disruptive technology
  - Galvanizes the power of the Internet
- Impact on communications similar to
  - SMTP on email or HTTP on web and browsing
- Leverages the infrastructure already in place
  - DNS
  - Supporting communication protocols
- Locate users, and determine their presence and capabilities
- Sets up, modifies and tears down sessions

What Did SIP Inherit?

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- URLs:
  - general references, recursive,
  - Can be embedded in web pages SIP: Dave@daveahlgren.com
- HTTP:
  - basic request/response format, status codes, . . .
  - proxies
  - CGI programming interface
  - email/SMTP:
  - Error codes, 404 file not found
- IP Addressing
  - MX & SRV mail server domain look up, load balancing, redundancy
- ENUM:
  - Translates phone numbers to SIP address via DNS
  - E.164

SIP Design Choices

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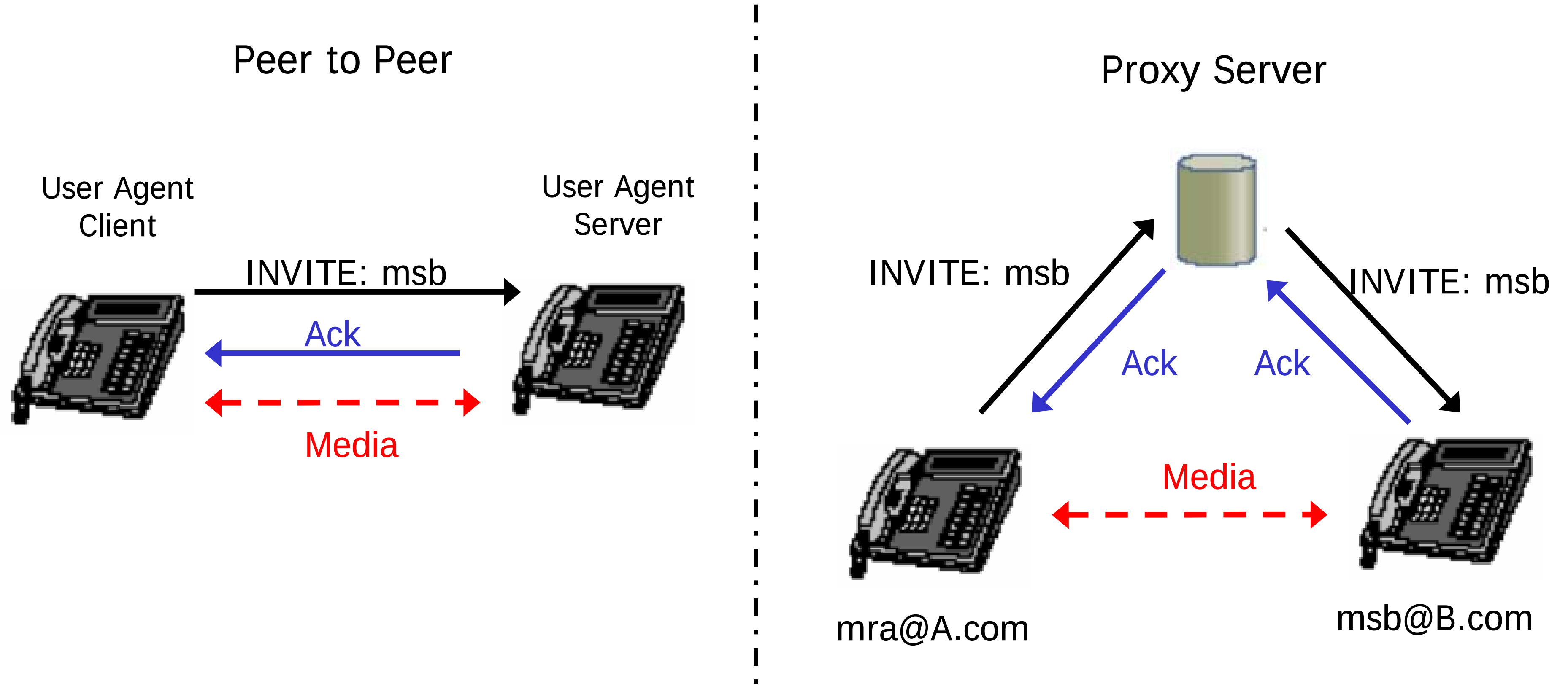
- Distributed architecture:
  - Smart end points and components
- Separate signaling:
  - Control, transport & media description:
  - Allows addition of new applications or media types
- Transport protocol neutrality:
  - Runs over reliable (TCP, SCTP)
  - Runs over unreliable (UDP) channels, with minimal assumptions
- Request based routing:
  - Direct routing for high performance or proxy-routed for control
- Extensibility:
  - Distributed architecture, smart UA, proxies, registrars
  - Flexible messaging formats

SIP is a Distributed Architecture



SIP is Not

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- SIP is not a turnkey application -
  - It is not Instant messaging, VoIP, or Video on demand.
  - It just standardizes the control protocol for those applications.
- SIP does not transport media
  - but does work with SDP
- SIP does not provide QoS
  - but can work with RSVP and RTP
- SIP does not provide Dir. Service or Authentication
  - but does work with RADIUS and LDAP

SIP Not a Transport

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- Independent of the packet layer
- Provides its own reliability mechanism.
  - Does not require a reliable datagram service
- Typically runs over UDP or TCP
  - Could run over, frame relay, ATM
  - Even carrier pigeons!
- Data packets most probably do not follow the same path as the SIP control packets

SIP Needs Other Protocols

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- SDP - Session Description Protocol
  - Describes the media content of the session
- RTP - Real Time Protocol
  - Carrier for voice or video
- SIMPLE– Proposed Standard
  - SIP for Instant Messaging and Presence Leveraging Extensions And a variety of other protocols
- RSVP, RADIUS (AAA) SOAP,WSDL, UDDI
- SIP needs alphabet soup to stay healthy!

Session Description Protocol (SDP)

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- Conveys information for a multimedia session
  - Media to use (codec, sampling rate)
  - Media destination (IP address and port number)
  - Session name and purpose
  - Contact information
- Note: indeed SDP is a data format rather than a protocol.

Real Time Transport Protocol (RTP)

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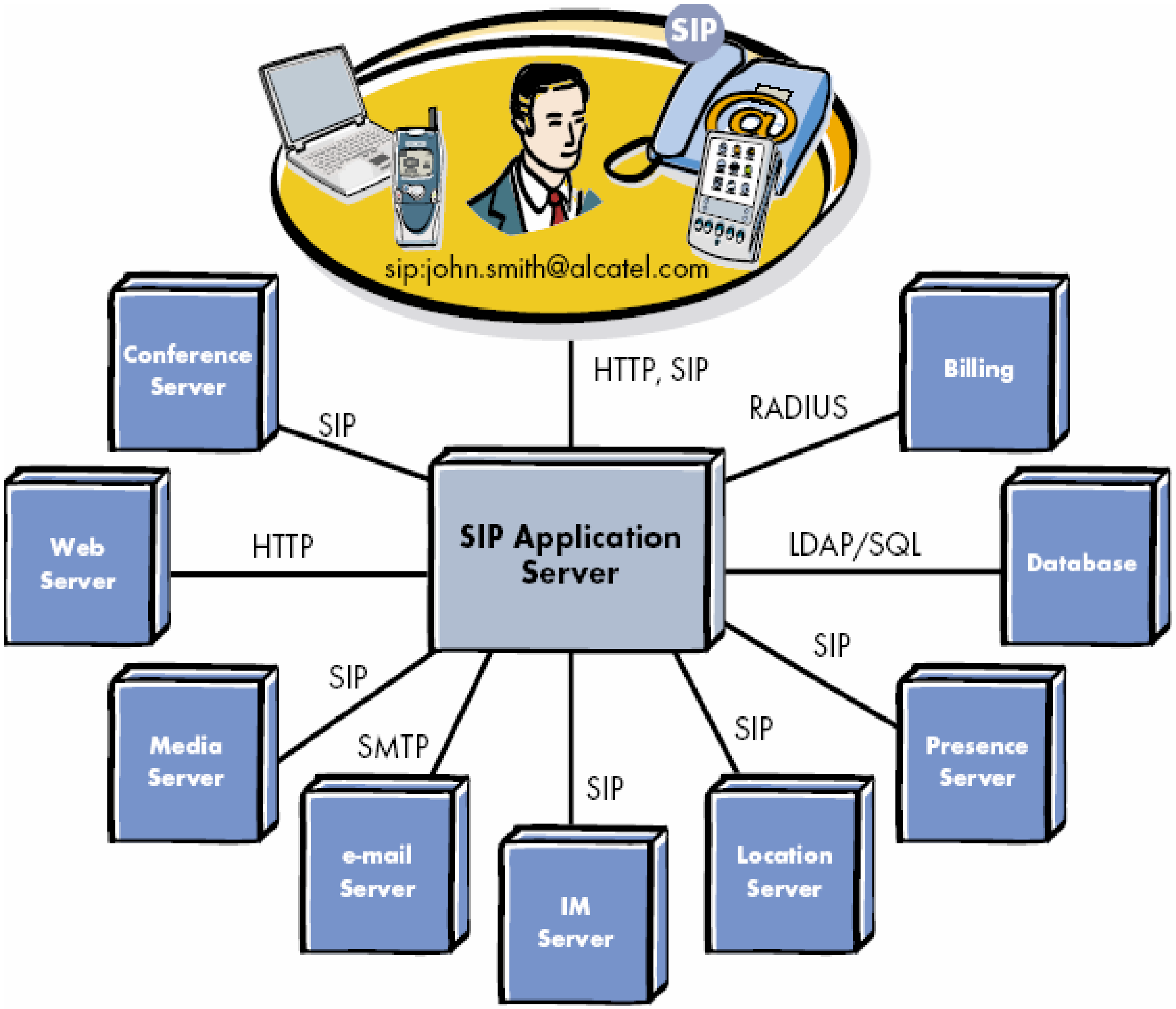
- Designed to be scalable, flexible
  - Media content type – Voice Video etc.
  - Sender identification – IP address
  - Synchronization - Random # identifying the source
  - Loss detection - Time stamp
  - Segmentation and reassembly - sequence #
  - Security (encryption)
- IP/RTP Message format
  - <Phy/MAC> <IP> <UPD> [<RTP header> <Media Content>]
  - [ ] = Payload

SIP Application Server



Alcatel

SIP Practical Issues

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- SIP's Prime competitor is H.323
  - It is closed protocol providing signaling and media transport
  - a 'Walled Garden', overly complex and fading
  - That is the good news!
- Peer to Peer communication, insufficient for commercial service
  - Proxy and registrar servers are required on the network side
- Firewall and NAT Transversal is problematic
  - Newer firewalls will recognize and pass SIP
- SIP does not support E911 calls or CALEA – lawful call intercept
  - The 3GPP SIP standards are solving some of these problems
- Session Border Controllers
  - Support CALEA and NAT but defeat some of the value of SIP

SIP Functionality

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User location:	Determination of the end system to be used for communication; Translate user name to their current network address
User availability:	Determination of the willingness of the called party to engage in communications; Presence – Your buddy list in IM
User capabilities:	Determination of the media and media parameters to be used; negotiation of features to be used between end points.
Session setup:	Establishment of session parameters at both called and calling party; "ringing",
Session management:	Adding, dropping, transfer and termination of sessions, modifying session parameters and invoking services.

Major Components of the SIP Environment

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User Agents	End-users devices or gateway to other networks - Cell phones, Multimedia handsets, PC's, PDA's; UAC - User Agent client initiates the message, UAS - User Agent server responds to the message
Registrar Servers	Data base of all User Agents within a domain. Retrieve participants IP address and forward to the Proxy Server
Proxy Severs	Accept Session Requests, Query Registrar Server for recipients' address
Redirect Servers	Allow SIP Proxy Servers to direct SIP sessions to external domains
Note - All 'servers' above may reside in the same hardware	

SIP Message

INVITE: sip:userB@aol.com m SIP/2.0
Via: SIP/2.0/UDP userA.yahoo.com
From: userA. <sip:userA@yahoo.com>
To: userB. <sip:userB@aol.com>
Call-ID: 7607607600@userA.yahoo.com
CSeq: 1 INVITE
Subject: Lunch today.
Content-Type: application/SDP
Content-Length: 182

Message Types

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INVITE	Initiates a call, changes call parameters (re-INVITE).
ACK	Confirms a final response by the final end point for INVITE.
BYE	Terminates a call.
CANCEL	Cancels INVITE (searches and “ringing”)
OPTIONS	Queries the capabilities of the other side.
REGISTER	Registers with the SIP proxy location service.
INFO	Sends mid-session information that does not modify the session state.
Note – Requests go from the client to the server	

Status Codes

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- Borrowed from HTTP.
  - x80 and higher codes avoid conflicts with future HTTP response codes
- Message from server to the client
- Provisional Codes - (1xx class)
  - indicate progress, Do not terminate SIP transactions
- Final Codes - (2xx, 3xx, 4xx, 5xx, 6xx classes)
  - Terminate SIP transactions.
- Status Codes
  - 1XX – information messages (100 – trying, 180 – ringing, 183 – progress)
  - 2XX – successful request completion (200 – OK)
  - 3XX – call forwarding, redirection (302 – temporarily moved, 305 – use proxy)
  - 4XX – client error, request failure (403 – forbidden)
  - 5XX – server error (500 – Server Internal Error, 501 – not implemented)
  - 6XX – global failure, busy, refused, not available (606 – Not Acceptable)

Basic Message Format

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SIP Control messages are human readable clear text

- START LINE Request or Status response
- HEADERS Fields Message attributes,
  - General format <name>:<value>

(Blank line marks end of SIP headers and beginning of body)

- BODY (CONTENT) Media specific parameters

Remember – The Media is transported separately

Message - START LINE

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- Each SIP message begins with a Start Line
- Conveys the message type and protocol version
- Can be a Request or Status Line.
  - Request Line   includes a Request URL of the destination user or service,
  - Status Line   includes the numeric Status-code and its associated textural phrase

Message - HEADERS Fields

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- Convey message attributes and modify message meaning.
- Similar in syntax and semantics to HTTP header fields
- To:, From:
- General format <field name> : <value>
- Headers can span multiple lines.
- Headers can appear multiple times in a message
- Via, Contact, Route and Request-Route

Message - BODY (CONTENT)

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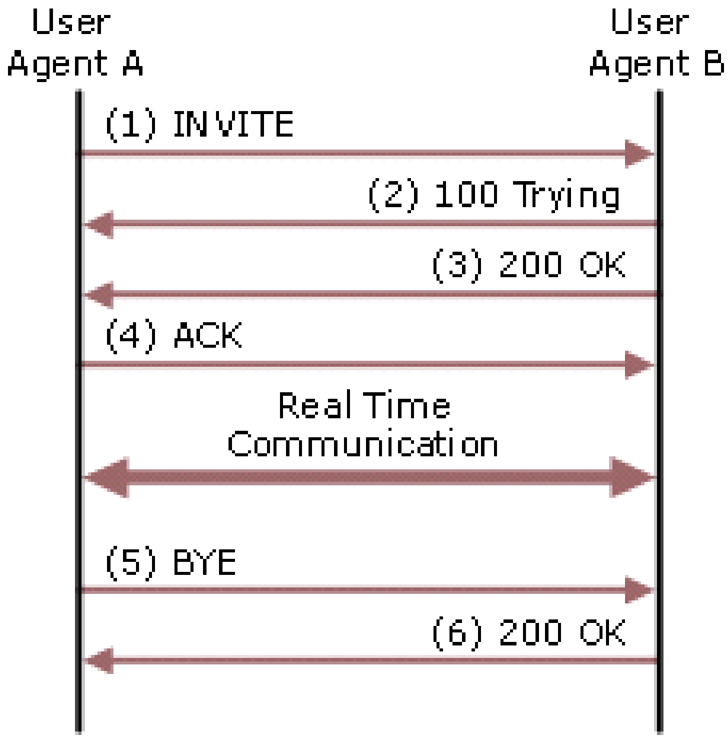
- Describes the session to be initiated, audio and video codec types, sampling rates etc
- Can appear both in request and in response messages
- Can contain any opaque information that relates to the session
- Body session types
  - SDP—see Session Description Protocol (SDP).
  - Multipurpose Internet Mail Extensions (MIME).
  - Others—to be defined in the IETF and in specific implementations.

Message - SAMPLE

Request Message line	Description
INVITE sip:userB@aol.com m SIP/2.0	Method type, request URI (of called party), SIP version. (ie. The Start Line)
Via: SIP/2.0/UDP	Address of previous hop. (ie. Begin Header)
userA.yahoo.com	(on next line)
From: userA. <sip:userA@yahoo.com>	User originating this request.
To: userB. <sip:userB@aol.com>	User being invited, as specified originally.
Call-ID: 7607607600@userA.yahoo.com	Globally unique ID of this call.
CSeq: 1 INVITE	Command sequence. Identifies transaction.
Subject: Lunch today.	Call subject and/or nature.
Content-Type: application/SDP	Type of body—in this case SDP.
Content-Length: 182	Number of bytes in the body.
Blank line marks end of SIP headers and beginning of body.	
v=0	Version of SDP. (ie. The start of Body)
o=userA 53655765 2353687637 IN IP4 128.3.4.5	Etc. Dependent on the content-type above

SIP Call Flow - Direct

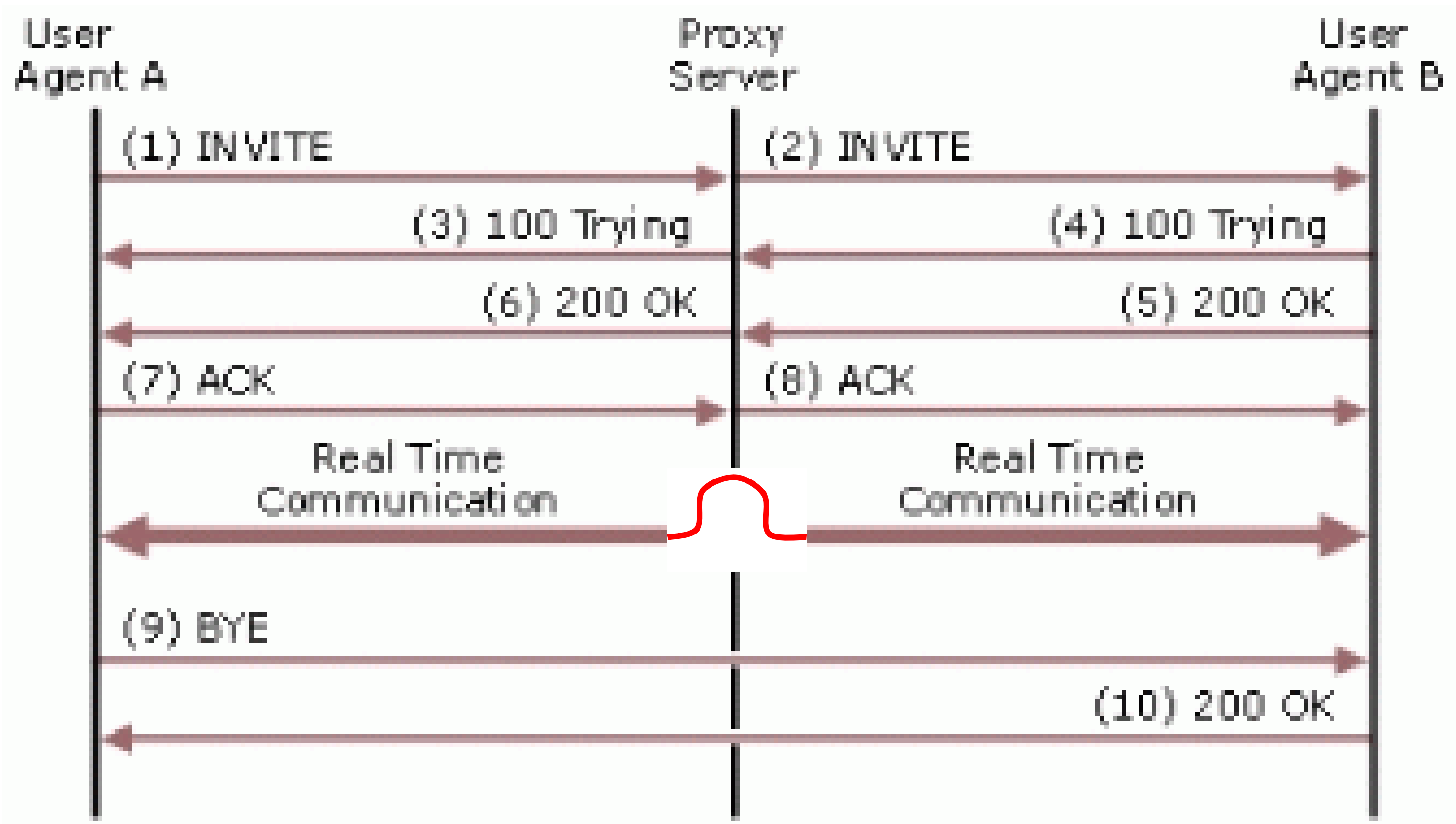
- The call flow for SIP sessions depends upon whether the SIP session is established directly between SIP user agents or whether a SIP server (proxy, registrar, or redirect) is located between SIP user agents.



- Note- The media transport may be on an entirely different path

SIP Call Flow with Proxy Server

- The proxy server is a communication midpoint, functioning as both a user server and as a user agent. When acting as a user server the proxy receives the SIP requests and forwards them on to the destination user agent and vice versa when acting as a user agent.



SIP Questions Now?

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- Control vs. Transport
- Extensions to other protocols
- User Agents, proxies, registrar
- Message formats
- Call flow

VoIP – Reinventing the Wheel

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- Voice over IP – Voice over broadband networks
  - Private networks
  - Public internet
- VoIP is a revolutionary technology
  - Potential to completely rework the world's phone systems.
- VoIP providers like Vonage have > 1,000,000 customers
- Major carriers like AT&T have VoIP calling plans in several markets around the United States
- Forrester Research Group predicts
  - 5 million VoIP U.S. households by the end of 2006.
- FCC is looking seriously at the potential ramifications of VoIP service.

VoIP Advantages

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- Lower cost
- Increased functionality

- More services, more user control
- Calls routed to where ever you plug in
- Take your phone where ever you go. you at home or on a trip
- Get the same services at the office, home or away
  - Dial extensions
  - Get Voice mail
- Join multiple offices, provide unified services
  - Pbx's, call centers agents can be located any where on the web

What services do you get?

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- Standard Features without extra cost
  - Caller ID
  - Call waiting
  - Call transfer
  - Repeat dial
  - Return call
  - Three-way calling
- Advanced Features controlled by user
  - Forward the call to a particular number
  - Send the call directly to voicemail
  - Give the caller a busy signal
  - Play a "not-in-service" message
  - Send a specific caller to a funny rejection

Making the VoIP Connection

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- ATA – Analog Telephone Adaptor
  - The most common VoIP connection is via an ATA. It connects between a standard phone and your Internet connection and provides VoIP service. The ATA converts the analog signal from your traditional phone and converts it into digital data for transmission over the Internet.
- IP Phones –
  - Look just like normal phones with a handset, cradle and buttons. IP phones have an RJ-45 Ethernet connector instead of having the standard RJ-11 phone connectors and connect directly to your router. Some IP phones must be connect to a dedicated network
- Soft Phone –
  - One of the easiest way to use VoIP. Down load free software and you can make free world wide computer to computer voice calls. All you need is a microphone, speakers and an Internet connection
- Generally, once connected, you can move from place to place and make calls.
  - All you need is an Internet connections.,

Product Examples

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ATA

Adaptor

$60

Cisco

IP Phone

$900

X Lite

Soft Phone

Free

And VoIP can be Invisible

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- Computers are embedded everywhere
- “VoIP” technology will appear in
  - Internet appliances
  - home security cameras, web cams
  - 3G mobile terminals
  - fire alarms and building sensors
  - chat/IM tools
  - interactive multiplayer games
  - 3D worlds: proximity triggers call

VoIP Architectures

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Features/Functions	**SIP**	**H.323**	**Megaco/MGCP**
Multiple domains	x	?	–
Third-party control	x	–	single-domain
Multimedia	x	fixed	set not likely
End system control	x	x	–
Extensible	x	?	limited
Generic events	x	–	–
CGI scripting	x	–	–
JAVA servlets	x	–	–

VoIP Codecs – Bandwidth Requirements

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Codec	Algorithm	Kb/s*	MOS**
G.711	PCM	64	4.1
G.726	ADPCM	32	3.85
G.729	CS-ACELP	8	3.92
G723.1	ACELP	5.3	3.56

* Kb/s can increase by ~1.5-2 x in wireless networks

** The voice transmission quality is measured by the MOS (Mean Opinion Score),

NAT Transversal

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- SIP end points behind Firewalls/NAT do not have public IP address
- There are at lease 4 types of NATs – Network Address Translation
  - Full Cone – port is open
  - Restricted Cone – port is open to destination IPs
  - Port Restricted Cone – open to destination IP – Port pairs
  - Symmetric - only the destination computer can respond
- NAT Transversal Gateways use NAT discovery techniques.
  - UPnP – Universal Plug n Play, Microsoft,
  - External Query – Client asks an external server
  - STUN – Simple Transversal of UDP through NAT
    - gets IP/port map + NAT type, Needs 4 tests
  - Symmetric NAT require RTP media stream to go thru gateway

VoIP Network Performance Considerations

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- Latency – Delay of packet delivery
  - 250 ms round trip max, 150 ms one way
  - SLA – Major backbone providers – 45 – 65ms
  - Does not include ISP or local LAN
- Jitter – Variations in delay
  - VoIP endpoints (IP phones, ATAs) have jitter buffers
  - Buffers add delay, 100 ms max
  - Major backbone providers – 0.5 – 2ms, up to 10 ms peak
- Packet Loss –
  - Excess network traffic esp. on wireless networks
  - Use of UDP, packet order errors
  - 1% loss on G.711 codec is significant, others tolerate less

SIP and QoS Control

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- SIP does not provide QoS support.
- SIP can requests QoS via the notion of preconditions.
  - ie. ask that resources are made available before the phone rings.
  - Invitations might indicate in SDP that QoS assurance is mandatory
  - All setup should only proceed after satisfying the preconditions
- SIP extension method (COMET) indicates the success or failure of the preconditions.

QoS Protocols

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- RSVP – Resource Reservation Protocol
  - Request priority service from each router on the path
  - Tries to emulate a traditional ‘circuit’
  - Provides high level service and feed back to the QoS application
  - ‘Circuit’ is built and torn down for each connection
- DiffServ – Differentiated Services
  - Two service level or traffic classes per hop
  - Expedited Forwarding – highest quality but drop excess traffic
  - Assured Forwarding - excess traffic is delayed
  - Marks traffic at ingress and un-marks at egress points
  - Use values in the DS byte (TOS)
  - Effective method for aggregating CBR streams into fixed pipes
- MPLS – Multi-protocol Label Switching
  - Similar to DiffServ, prepares/reserves the network for QoS

Questions – VoIP

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- Applying VoIP
- Advantages
- Services
- Network requirement
- QoS

Attachments

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- SIP and VoIP Reference and Glossary

Thank You

Please contact me for a copy of these slides

David Ahlgren

760 846 0357

Dave@drahlgren.com

