SIP and the IETF Vision for IP Telephony Deployment

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Overview

- VoIP service models
- the IETF VoIP architecture
- the Session Initiation Protocol (SIP)
- challenges on the horizon:
  - emergency services
  - instant messaging & presence
  - generic event notification
  - integration with 2G mobile (GSM, CDMA)
  - next-generation wireless (3GPP, 3GPP2, MWIF, …)
Internet Telephony Service Models

- Internet “PBX”
- Internet Centrex
- Internet Carrier

same basic equipment, but size of gateway varies
Internet PBX

- SIP proxy server
- IP
- ISP
- Ethernet
- T1, ISDN (BRI/PRI) or analog lines
IP Centrex

Chatterbox Cafe

Ralph’s Pretty Good Grocery

Heads Up Barber

ISP

Internet

PSTN
IP Carrier

SS7
MGCP
H.248
SIP
IP
010

IP Carrier Diagram:
- San Francisco: PacificBell
- Denver: Qwest
- LA: PacificBell
- Chicago: MCI
- New York: Verizon
- Montreal: Bell Canada
- Dallas: BellSouth
- El Paso: BellSouth
IETF VoIP Protocol Architecture

Languages/APIs
- JAIN
- CPL
- voiceXML
- Parlay
- servlets
- sip-cgi

Directory/Discovery
- DNS/enum
- LDAP
- TRIP
- SLP

Signaling
- SIP
- MGCP
- H.248
- PINT
- SPIRITS
- RTSP

peer-to-peer
master-slave

QoS
- DiffServ
- IntServ

Transport
- RTP
- SCTP
- TLS
IETF VoIP Protocols & APIs

Most protocols are re-used —→ core

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Description</th>
<th>Protocol</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>session setup, services</td>
<td>CPL</td>
<td>XML-based language</td>
</tr>
<tr>
<td>MGCP</td>
<td>gateway control</td>
<td>sip-cgi</td>
<td>SIP-based scripts</td>
</tr>
<tr>
<td>SDP</td>
<td>describe multimedia sessions</td>
<td></td>
<td></td>
</tr>
<tr>
<td>RTP</td>
<td>multimedia transport</td>
<td></td>
<td></td>
</tr>
<tr>
<td>TRIP</td>
<td>find nearest gateway</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
## IETF Protocol Reuse

<table>
<thead>
<tr>
<th>protocol</th>
<th>designed for</th>
<th>VoIP use</th>
</tr>
</thead>
<tbody>
<tr>
<td>RTSP</td>
<td>streaming media</td>
<td>voicemail</td>
</tr>
<tr>
<td>DNS</td>
<td>name lookup</td>
<td>E.164 mapping</td>
</tr>
<tr>
<td>SCTP</td>
<td>reliable transport</td>
<td>ISUP transport</td>
</tr>
<tr>
<td>PGP</td>
<td>email</td>
<td>call authentication</td>
</tr>
<tr>
<td>MIME</td>
<td>email</td>
<td>signaling info</td>
</tr>
<tr>
<td>SDP</td>
<td>multicast sessions</td>
<td>SIP, MGCP</td>
</tr>
</tbody>
</table>
Protocol “Holes”

- “tight” session control for conferences
  - admission control
  - multicast key distribution
  - advanced capability negotiation
- scalable authentication for individuals
- cross-provider QoS: primarily a business problem
IETF VoIP Architecture Characteristics

- universal identifier *user@domain*: SIP URL = email = NAI
- separation of transport of services
- media-neutral, including beyond audio and video
- emphasis on user-programmable services
- web integration: content, mutual referral
- integration with IM and presence
## Internet QoS Architecture

<table>
<thead>
<tr>
<th>scheduling</th>
<th>flow</th>
<th>class</th>
</tr>
</thead>
<tbody>
<tr>
<td>admission flow</td>
<td>RSVP (int-serv)</td>
<td>BB? RNAP?</td>
</tr>
<tr>
<td>class</td>
<td>–</td>
<td>diff-serv</td>
</tr>
<tr>
<td>scheduling</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
SIP Overview

- protocol for establishing, modifying, tearing down (multimedia) sessions
- IETF Proposed Standard since March 1999
- multimedia = audio, video, shared applications, text, . . .
- also used for “click-to-dial” (PINT wg) and possibly Internet call waiting (SPIRITS wg)
- to be used for PacketCable Distributed Call Signaling
- to be used for Third-Generation Wireless (3GPP, 3GPP2)
# SIP Components

<table>
<thead>
<tr>
<th>Entity</th>
<th>Does</th>
<th>Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td>proxy server</td>
<td>forward calls</td>
<td>firewall controller, “call router”</td>
</tr>
<tr>
<td>redirect server</td>
<td></td>
<td>“application server”</td>
</tr>
<tr>
<td>user agent</td>
<td>end system</td>
<td>SIP phone, gateway, “softswitch”</td>
</tr>
<tr>
<td>registrar</td>
<td>location mgt.</td>
<td>mobility support</td>
</tr>
</tbody>
</table>

Roles are changeable, on a request-by-request basis
SIP Example: Redirection

1. INVITE henning@ieee.org
2. 302 Moved temporarily
   Contact: hgs@columbia.edu
3. ACK henning@ieee.org
4. INVITE hgs@columbia.edu
5. 200 OK
6. ACK hgs@columbia.edu
SIP Example: Proxying

1. INVITE henning@columbia.edu
2. 200 OK
3. INVITE hgs@play
4. 200 OK
5. 200 OK
6. ACK hgs@play
7. media stream
SIP Forking Proxies

INVITE sales@macrosoft.com
carol@c.macrosoft.com

INVITE bob@b.macrosoft.com

200 OK

INVITE carol@c.macrosoft.com

CANCEL bob@c.macrosoft.com

BYE carol@c.macrosoft.com

200 OK

macrosoft.com

a.wonderland.com

INVITE sales@macrosoft.com

ACK

INVITE carol@c.macrosoft.com

200 OK

Cancel bob@c.macrosoft.com
## SIP syntax

### Request

<table>
<thead>
<tr>
<th>Method</th>
<th>URL</th>
<th>SIP/2.0</th>
<th>SIP/2.0 status reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>Via:</td>
<td>SIP/2.0</td>
<td>protocol host:port</td>
<td></td>
</tr>
<tr>
<td>From:</td>
<td>user <a href="">sip:from_user@source</a></td>
<td></td>
<td></td>
</tr>
<tr>
<td>To:</td>
<td>user <a href="">sip:to_user@destination</a></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call-ID:</td>
<td>localid@host</td>
<td></td>
<td></td>
</tr>
<tr>
<td>CSeq:</td>
<td>seq# method</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Content-Length:</td>
<td>length of body</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Content-Type:</td>
<td>media type of body</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Header:</td>
<td>parameter ;par1=value ;par2=&quot;value&quot; ;par3=&quot;value folded into next line&quot;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>V=0</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>o= origin_user timestamp</td>
<td>IN</td>
<td>IP4</td>
<td>host</td>
</tr>
<tr>
<td>c=IN</td>
<td>IP4</td>
<td>media destination address</td>
<td></td>
</tr>
<tr>
<td>t=0 0</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>m= media type</td>
<td>port</td>
<td>RTP/AVP</td>
<td>payload types</td>
</tr>
</tbody>
</table>

### Response

<table>
<thead>
<tr>
<th>method URL</th>
<th>SIP/2.0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status</td>
<td>reason</td>
</tr>
</tbody>
</table>

### Message

- Blank line
- \( V=0 \)
- \( o= \) origin_user timestamp timestamp \( \text{IN} \) IP4 host
- \( c=\text{IN} \) IP4 media destination address
- \( t=0 0 \)
- \( m= \) media type port RTP/AVP payload types
SIP Advanced Features

- forking
- extensibility: new headers, methods, bodies
- security: web-like, PPP/CHAP or PGP
- multicast-capable
- support for personal, session, terminal, service mobility
- caller preferences: direct calls based on properties
## SIP Mobility

<table>
<thead>
<tr>
<th>Component</th>
<th>Description</th>
<th>Request(s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>terminal</td>
<td>cross-provider</td>
<td>REGISTER, re-INVITE</td>
</tr>
<tr>
<td>personal</td>
<td>different terminals, same address</td>
<td>REGISTER</td>
</tr>
<tr>
<td>service</td>
<td>different terminals, same services</td>
<td>upload</td>
</tr>
<tr>
<td>session</td>
<td>move sessions across terminals</td>
<td>REFER</td>
</tr>
</tbody>
</table>
SIP Personal Mobility

alice@host.columbia.edu
(also used by bob@columbia.edu)

tel:12128541111

tel:12015551234

alice@host.columbia.edu
(also used by bob@columbia.edu)

tel:12128541111

tel:12015551234

alice17@yahoo.com

alice@columbia.edu

7000@columbia.edu

Alice.McBeal@columbia.edu

columbia.edu

yahoo.com
Example SIP System
SIP-Based Telephony Services

- conferencing: “dial-in”, “dial-out”
- forwarding: basic SIP
- ACD: proxy, no protocol extensions
- call transfer: REFER extension
- DTMF transport: in RTP, not SIP
- billing: in resource reservation, (mostly) not SIP
Current SIP efforts

- SIP to Draft Standard
- QoS and security preconditions
- inter-domain AAA and billing
- session timer for liveness detection
- early media (PSTN announcements)
- SIP for presence / instant messaging
- SIP-H.323 interworking
- reliable provisional responses
- DHCP configuration for finding SIP servers
- SIP for firewalls and NATs
- caller preferences
- services (transfer, multiparty calls, home)
- ISUP carriage
SIP Emergency Services

• need
  – emergency address
  – find nearest PSAP
  – PSAP determines caller location

• cannot just rely on gateway calling 911

• generally, allow devices to be location-aware ("what time is it where I’m about to call?" “call pizza parlor”)

• offers new opportunities: database access, video, measurements, accessibility, …
SIP Emergency Services

INVITE sip:911
GPos: 42 21 54 N 71 06 18 W
GL: S3.USA.45420.1910

user database (location, room number, ...)

first-hop switch

INVITE sip:911
GL: S3.USA.45420.1910

location announcement for each wire

GPS

RADIUS or private protocol

customer database (names, addresses)

gps <--> civil translation database
SIP Bake-Off

• takes place every four months, 5th at Pulver.com August 2000
• 45 organizations from 11 countries
• about 50-60 implementations:
  – IP telephones and PC apps
  – proxy, redirect, registrar servers
  – conference bridges
  – unified messaging
  – protocol analyzers
• first IM/presence interop test
• emphasis on advanced services (multi-stage proxying, tel URLs, call transfer, IVR, …)
The Dangers of VoIP

- focus on single service: voice, fax, ...
- PSTN: service orientation ↔ Internet: neutral transport
- APIs as least common denominator across POTS, ISDN, SS7 → 100-year old functionality
- carbon-copy replication of existing services
- terminology overload
Differences: Internet Telephony ↔ POTS

- separate control, transport (UDP) ↔ no triangle routing
- separate connectivity from resource availability
- separate services from bit transport
- datagram service ↔ less bootstrapping
- in-band signaling ↔ higher speed
- features “network” → end system: distinctive ringing, caller id, speed dialing, number translation, … ↔ scaling
- features: intra-PBX = inter-LATA and general
- protocols: user-network = network-network signaling
PSTN Legacies to Avoid

- E.164 numbers – might as well wear bar codes
- tones (e.g., failure indications)
- in-band signaling (DTMF)
- systems with user interface knowledge (12 keys, voice)
- voice-only orientation (e.g., MGCP/Megaco)
- integration of bit transport and services
- service-specific billing
- trigger model for service creation
- trusted networks without crypto authentication

→ confine PSTN knowledge to edge of network
Replication of Existing Services

• “user is familiar with PSTN services”
• but how many users actually know how to use call transfer or directed pick-up?
• user interface is often just legacy of key systems or other ancient technology
• avoid binding of identifiers to devices – call person or group of people, regardless of location
• instead, model desired behavior
• single-server features don’t need standardization
• find general mechanisms (e.g., REFER for three-party calls and various call transfers)
Terminology Overload

Invasion of the meaningless technical-sounding terms, attempting to familiar mimic PSTN boxes:

- CO switch $\rightarrow$ soft switches = gateway + SIP UA + ?
- SCP $\rightarrow$ application servers = proxy? web server? media server?
- PBX $\rightarrow$ Internet PBX = proxy? + gateway?
- ...

Temptation: new name $\rightarrow$ new protocols, APIs, . . . – the old box boundaries don’t necessarily make sense!
It’s That Simple…

We really only have a few basic components:

- PSTN gateway, with some combination of FXO/FXS
- SIP proxy/redirect/registrar servers (or H.323 gatekeepers)
- SIP user agents (or H.323 terminals): PCs, phones
- media storage servers
- DNS, directory, web, email, news, … servers
Invisible Internet Telephony

VoIP technology will appear in . . .

- Internet appliances
- home security cameras, web cams
- 3G mobile terminals
- fire alarms
- chat/IM tools
- interactive multiplayer games
The Largest Signaling Network is Not Running SS7

- AT&T: 280 million calls a day
- AOL: 110 million emails/day, total about 18 billion/day
- total $> 1$ billion instant messages a day (AOL: 500 million)
- signaling effort of call $\approx$ IM
Signaling and Events

Signaling: “do this” (push) – Events: “this just happened”
Commonalities between Signaling and Events

• presence is just a special case of events: “Alice just logged in” ≈ “temperature in boiler exceeds 300° F”

• need to *locate* mobile end points

• may need to find several different destinations ("forking")

• same addressing for users

• presence often precursor to calls

• likely to be found in same devices

• events already in VoIP: message alert, call events
SIP as a Presence Platform

- requires minimal extensions to SIP: **SUBSCRIBE** to ask to be alerted, **NOTIFY** when event occurs

- **MESSAGE** for sending text messages (“IM”)

- true “chat” is voice (+ video)

- services such as reaching mobile phone while in meeting
Events: SIP for Appliances

(SIP user agent)

SIP proxy (RGW)

- SUBSCRIBE door@alice.home.net
- NOTIFY alice@work.com
- DO light@alice.home.net
- INVITE camera@alice.home.net

(Work with Telcordia)
# Programmable Internet Telephony

<table>
<thead>
<tr>
<th>Feature</th>
<th>APIs</th>
<th>servlets</th>
<th>sip-cgi</th>
<th>CPL</th>
</tr>
</thead>
<tbody>
<tr>
<td>Language-independent</td>
<td>no</td>
<td>Java only</td>
<td>yes</td>
<td>own</td>
</tr>
<tr>
<td>Secure</td>
<td>no</td>
<td>mostly</td>
<td>no, but can be</td>
<td>yes</td>
</tr>
<tr>
<td>End user service creation</td>
<td>no</td>
<td>yes</td>
<td>power users</td>
<td>yes</td>
</tr>
<tr>
<td>GUI tools w/portability</td>
<td>no</td>
<td>no</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>Call creation</td>
<td>yes</td>
<td>no</td>
<td>no</td>
<td>no</td>
</tr>
<tr>
<td>Multimedia</td>
<td>some</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
</tr>
</tbody>
</table>

Example: integration with iCal → automatically export personal calendar to call handling
Third-Generation Wireless

- goal: 144 kb/s moving, 384 kb/s stationary, 2 Mb/s indoors
- based on GSM or wideband CDMA
- push IP to the hand set
- SIP as signaling system for voice calls in 3GPP
Conclusion

- basic IETF-based architecture in place
- SIP as foundation for services – see http://www.cs.columbia.edu/sip
- extensions to mobility, emergency services, ... in progress
- first (and last?) chance to recover from 120 years of legacy
- avoid replication of PSTN on packets
- most VoIP applications won’t look like a telephone
- opportunities in emergency services, mobile, event notification