State of SIP
U of MN
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By: Tim Kraskey

Who is Tim Kraskey!
- Entrepreneur
- Educator
- Marketing and Operations Executive
- Angel, Venture Capitalist and Investor

Who is Calabrio, Inc?
- Formed in 2007 as the software products division of Spanlink Communications, Inc.
- Develops and distributes customer interaction software
- **Calabrio Unified Interaction Suite** consists of:
  - Cisco Agent Desktop
  - Cisco Supervisor Desktop
  - Cisco Workforce Optimization
  - Calabrio Workforce Management
  - Calabrio Quality Management
- Distribution through Cisco OEM and Channel Partners
- Software on more than 500,000 desktops

Macro Market Trends
- Legacy (TDM) to IPTel Migration
  - Gartner by 2008 New Sales of IPTel = 97%
  - SIP is a key driver
- IP Contact Centers and advanced applications lag (2 Years) IPTel migration
- Drive to Integrated (Unified) Application Solutions

Market Problem
- Integrating desktop applications is very complex
- No seamless integration of best practice and processes that drive relationship improvements
- SIP and SOA help but are they standards?
  - Many extensions to the standards
- VoIP and Contact Center Applications = “Science Project”
**Problem: “Customer Interaction Networks” Require a “Suite” of Integrated Applications**

- Corporate HQ
- Enterprise Information
- Contact Center
- Voice and Data Center

**YankeeGroup Study: Problems with a Typical Agent Desktop**

- More than 65% of contact center agents use three or more applications.
- More than 25% use five or more applications.
- 70% say they waste time switching between applications.

**Market Opportunity**

- “All Customer Interaction Apps” up for grabs
- Cisco is the No. 1 IPTel leader today!
- Microsoft is coming!
- Market size near $4B
- Applications for the Virtual Agents and Knowledge Workers

**IP Adoption - Enterprise and Contact Center**

- Total Agent Seats: 5% Annual Growth
- TDM ACD: CAGR: -6%
- IP ACD: CAGR: +39%

**SIP Architecture - Ref: Cisco**

[Diagram showing SIP architecture related to Cisco's implementation]
SIP Interoperability and Extensions

Slides Courtesy of Edwin E. Mier, CEO, Mier Consulting, LLC
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SIP Interop: Different Views

- Summary results of third annual survey of SIP-implementing vendors
- Who are the SIP interop leaders?
- SIP-interop status of seven leading IP-PBX vendors

Where SIP Matters: Key Product Categories

- IP PBXs and call controllers
- Gateways
- SIP Endpoints (incl softphones, wireless)
- SIP application servers (UC, conf, collab)
- SIP trunks (IP-PBX <-> service provider)

Mier Survey says …

- Survey emailed to ~ 85 vendors
- 36 complete responses received by deadline (incompletes, duplications were eliminated)
- About 40 percent of the SIP vendor community represented (all product categories)
- SIP-based carriers were not included
- Vendors answered probing questions about their SIP implementation, interoperability and plans

Issues Asked About!

- Features: What’s standard SIP? What’s not?
- How solid are the SIP specs?
- Are SIP products interoperable today?
- What are the “most interoperable” SIP features?
- Are things getting better, re: SIP interop?
SIP Features, Extensions and Interop

- Prospects for multi-vendor interoperability
  - Solid SIP RFC features – Excellent
    - ~ 20 features matter the rest are subjective
  - "Feature codes" – Good, but vendor specific
  - Proprietary SIP extensions – Poor (w/o collaboration)
    - All vendor have them

State of SIP Specs

- Vendors asked to rate “the state of current SIP specs, from all sources …”
- “… for implementing 24 features and capabilities
- Using a 1 to 5 rating scale
  5 = complete, solid, clear, stable, unambiguous
  1 = minimal to no standardization yet; or incomplete or ambiguous; needs a lot of work

State of SIP Specs – Most Solid

- Basic dozen phone features
- Voice mail
- Caller ID
- Presence monitoring
- Propagation
- Audio conferencing
- Video conferencing
- SIP trunking

State of SIP Specs – Least Solid

- Drag/drop file transfer
- Call Ctrl processes
- Whiteboarding
- Remote config of SIP phones
- Web collaboration
- "Advanced" features
- CDR records

State of SIP Specs – So-so

- Firewall traversal
- Realtime monitoring
- Call control security
- RTP voice encryption
- NAT traversal
- Scheduled audio conf
- Find-me / follow-me

State of SIP Specs – Bottom Line

- In only a few areas is there widespread agreement the specs are solid and complete (basic dozen phone features, voice mail, presence, ad hoc audio and point-point video conferencing)
- "Advanced" applications and phone features are rated generally as “there are some specs, but a lot more detail is needed”
SIP Product Interoperability

- Vendors asked to "Assess the state of inter-vendor SIP-product interoperability …"
- Given 8 environments
- And using a 1 to 5 rating scale
  - 5 = Plug-and-play, full-featured interoperability
  - 1 = No chance of any meaningful interoperability without a lot of work and tweaking

SIP Product Interoperability

4.2
3.7
3.7
4.5
1
2
3
4
5

- For "basic telephony features," w/ SIPIT and direct testing between 2 vendors
- For SIP trunking, IP-PBX to carrier, 2 days testing and adjustment
- For "basic telephony features," no prior collaboration between 2 independent vendors
- For "advanced telephony features," w/ SIPIT and direct testing

SIP Product Interoperability

3.1
2.7
2.2
3.3
1
2
3
4
5

- For SIP trunking, IP-PBX to carrier, based on independent implementation of SIP Forum’s spec
- For "collab and multimedia applns," w/ SIPIT and direct testing between 2 independent vendors
- For "enhanced telephony features," no prior collaboration between 2 independent vendors
- For "collab and multimedia applns," no prior collaboration between 2 independent vendors

SIP Product Interop – Bottom Line

- Interop prospects are now good for "basic" telephony features, even with no prior collaboration between vendors
- Good chance of SIP-trunking interop … after a couple days of shake-down testing
- All else, users should insist on SIPIT or direct collaboration/testing between 2 vendors

Most interoperable SIP endpoints

- Top 5 SIP endpoint vendors, based on how many other vendors claim interop with:
  - Polycom
  - Cisco phones w/ SIP load
  - CounterPath / Xten / eyeBeam softphone
  - Grandstream
  - Snom

Other very interoperable SIP gear

- Many vendors also claim interop with:
  - Hitachi wireless
  - Linksys
  - Quintum gateways
  - Aastra
  - Thomson
Most interoperable SIP trunks

- Leading SIP-trunk-accessible services, based on how many vendors claim interop:
  - AT&T (Flex Reach)
  - Verizon (Verizon Business, MCI)
  - cBeyond
  - AGN Networks
  - Bandwidth.com

IP-PBX SIP Support

- A comparative look at the SIP status, claims and plans of seven IP-PBX vendors:
  - Alcatel-Lucent
  - Avaya
  - Cisco
  - Mitel
  - NorTel
  - Siemens
  - 3Com

Alcatel-Lucent SIP Support

- Main SIP-supporting platform(s): OmniPCX Enterprise, and OmniTouch Unified Comms applns (media) server
- Is SIP primary call control? Optional in PBX, along with H.323. Native SIP in app server.
- Vendor offers SIP phones? No
- SIP standard RFC features: 16 (100%)
- SIP draft-based features: 0 (0%)
- SIP proprietary headers or features codes: 0 (0%)

Avaya SIP Support

- SIP-call Security: TLS, no secure RTP, authentication
- Extent of validated SIP interoperability:
  - 3rd-party SIP phones: 17 vendors
  - Carrier services via SIP trunks: 5 (Currently supporting all the SIP Forum’s IP-PBX / Service Provider Interop recommendations for IP-PBX’s labeled as MUST)
  - Applns server: Meeting Exchange is SIP based
Cisco SIP Support

- Main SIP-supporting platform(s): Unified Comms Mgr (nee CallManager)
- Is SIP primary call control? Yes, and/or SSCP. MGCP to gateways, and H.323 via protocol gateway.
- Vendor offers SIP phones? Yes (half-dozen + soft)
- SIP standard RFC features: 90 (50%)
- SIP draft-based features: 20 (10%)
- SIP proprietary headers or features codes: 70 (40%)

Cisco SIP Support

- SIP-call Security: TLS, secure RTP, and authentication
- Extent of validated SIP interoperability:
  - 3rd-party SIP phones and gateways: per RFC 3261
  - Carrier services via SIP trunks: No specific carriers or service providers cited
  - Applns server(s): Half-dozen applns servers; all are accessible via SIP trunks

Mitel SIP Support

- Main SIP-supporting platform(s): Mitel 3300 ICP
- Is SIP primary call control? It can be, and/or MiNet proprietary VoIP call protocol.
- Vendor offers SIP phones? Yes (half-dozen models), which work w/ a dozen other vendors’ call controllers
- SIP standard RFC features: 12 (3%)
- SIP draft-based features: 1 (< 1%)
- SIP proprietary headers or features codes: ~300 (97%)

Mitel SIP Support

- SIP-call Security: no TLS, no secure RTP, authentication
- Extent of validated SIP interoperability:
  - 3rd-party SIP phones and gateways: 8 vendors (but Mitel’s SIP phones support 75 features, work with many vendor’s SIP call controllers)
  - Carrier services via SIP trunks: 5 service providers (and SIP trunks to dozen-plus other call controllers)
  - Applns server(s): Messaging and conference servers support SIP

Nortel SIP Support

- Main SIP-supporting platform(s): MCS 5100 applns server; working with CS 1000, CS 2000, CS 2100
- Is SIP primary call control? It can be, and/or Unistim proprietary VoIP call protocol; and H.323 support
- Vendor offers SIP phones? Yes, 4 models + soft, which work w/ Nortel’s call controllers
- SIP standard RFC features: ~45 (10%)
- SIP draft-based features: ~120 (30%)
- SIP proprietary headers or features codes: ~300 (60%)

Nortel SIP Support

- SIP-call Security: TLS and secure RTP (by call controller)
- Extent of validated SIP interoperability:
  - 3rd-party SIP phones and gateways: 9 vendors (Nortel’s SIP phones work with Nortel SIP-based call control)
  - Carrier services via SIP trunks: 1 service provider cited, SIP trunks to 4 other vendors’ call controllers
  - Applns server(s): Nortel’s MCS 5100/5200 is primarily a SIP-based conferencing and applns server
Siemens SIP Support

- Main SIP-supporting platform(s): New HiPath 8000, and OpenScape, a SIP-based applns server
- Is SIP primary call control? Yes, with MGCP support for gateways
- Vendor offers SIP phones? Yes, half-dozen models + softphone
- SIP standard RFC features: ~40 (40 %)
- SIP draft-based features: ~45 (45 %)
- SIP proprietary headers or features codes: ~15 (15 %)

Siemens SIP Support

- SIP-call Security: TLS, no secure RTP (planned)
- Extent of validated SIP interoperability:
  - 3rd-party SIP phones and gateways: 8 vendors (Siemens’ SIP phones work with several other vendors’ carrier-oriented SIP-based call controllers)
  - Carrier services via SIP trunks: None cited, testing based on SIP Forum SIP-trunking spec is underway
  - Applns server(s): Vendor’s OpenScape works with vendor’s call controllers, and Microsoft

3Com SIP Support

- Main SIP-supporting platform(s): VCX, and IBM System i IP Telephony
- Is SIP primary call control? Yes
- Vendor offers SIP phones? Yes, half-dozen models + softphone
- SIP standard RFC features: ~45 (10 %)
- SIP draft-based features: ~120 (30 %)
- SIP proprietary headers or features codes: ~300 (60 %)

3Com SIP Support

- SIP-call Security: No TLS, no secure RTP (both planned for 3Q07), authentication
- Extent of validated SIP interoperability:
  - 3rd-party SIP phones and gateways: 12 vendors (3Com’s SIP phones work with 2 other vendors’ SIP-based call controllers, supporting about 50 features)
  - Carrier services via SIP trunks: 2 service providers cited
  - Applns server: Applns server is also SIP-based

Review

- In which areas are SIP implementations most likely to operate … and not to interoperate?
- What sorts of features are being implemented as SIP extensions (feature codes, proprietary headers) and why?
- Will SIP extensions always be with us, or will most features become standardized over time?

Key Reference Sites

- IETF - http://www.ietf.org/
- SIP Forum - http://www.sipforum.org/