State of SIP
U of MN
10/26/07
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
Cisco is Winning the VoIP Race
per Synergy Research Group—Q4 CY06

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 Application

- **Customer** interacts with **Corporate HQ**
- **Voice and Data Center**
- **Unified Communicator**
- **Agent Desktop**
- **Centralized IT/ACD**
- **Business Analytics**
- **Centralized Storage**
- **Knowledge Base**
- **Backend Apps**

**Enterprise Information**
- **Find Info**
- **Schedule**
- **Consult**
- **Call Redirected**
- **Record**
- **Assist**
- **Up Sell**

**Outbound Workflows**
- **Supervisor Sees Call is too Long!**
- **Agent Can’t Find Answer Needs Help!**
- **Call Redirected**

**Branch 1**
- **Supervisor Desktop**
- **Coach**
- **Train**
- **Adherence**
- **Intra-Day Schedule**
- **Supervisor Sees Call is too Long!**
- **“Virtual” Supervisor**

**Branch 2**
- **Supervisor Desktop**
- **Report**
- **Intra-Day Schedule**
- **“Virtual” Supervisor**

**Contact Center**
- **Agent Desktop**
- **Agent Busy**
- **“Virtual” Agent Teams**
- **Agent Looking for Answer**

**Home**
- **Agent Desktop**
- **Record**
- **Schedule**
- **Mobility**
- **Assist**

**Virtual Agent Teams**
- **Subject Matter Experts**

**Unified Communicator**
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.

Source: Yankee Group, 2006
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%

85% of CC < 100 Seats in Size

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<tr>
<th>Year</th>
<th># WW Agent Seats</th>
<th># WW IP Agent Seats</th>
<th># WW TDM Agent Seats</th>
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85% of CC < 100 Seats in Size
SIP Architecture - Ref: Cisco
SIP Architecture - Ref: Cisco
SIP Interoperability and Extensions

Slides Courtesy of Edwin E. Mier, CEO, Mier Consulting, LLC
Kunkletown, PA
emier@mierconsulting.com
610-295-5132
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" telephony features: 4.4
- Voice mail: 3.9
- Caller authentication: 3.9
- Presence monitor'g, propagat'n: 3.6
- Audio conf ad hoc: 3.6
- Video conf point-point: 3.6
- SIP trunking: 3.5
State of SIP Specs – Least Solid

- Drag/drop file transfer: 2.3
- Call Ctr processes: 2.4
- White-boarding: 2.5
- Remote config of SIP phones: 2.7
- Web collaborat'n: 2.8
- "Advanced" features: 3.0
- CDR records: 3.0
State of SIP Specs – So-so

- Firewall traversal: 3.2
- Realtime monitoring: 3.2
- Call control security: 3.2
- RTP voice encryption: 3.3
- NAT traversal: 3.3
- Scheduled audio conf: 3.3
- Find-me / follow-me: 3.3
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

For "basic telephony features," w/ SIPIT and direct testing between 2 vendors
For SIP trunking, IP-PBX to carrier, 2 days testing and adjustm't
For "basic telephony features," no prior collaborat'n between 2 independ vendors
For "advanced telephony features," w/ SIPIT and direct testing
SIP Product Interoperability

- For SIP trunking, IP-PBX to carrier, based on independent implement'n of SIP Forum's spec: 3.3
- For "collab and multimedia applns," w/ SIPIT and direct testing between 2 independ vendors: 3.1
- For "advanced telephony features," no prior collaborat'n between 2 independ vendors: 2.7
- For "collab and multimedia applns," no prior collaborat'n between 2 independ vendors: 2.2
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
  -- Cisco
  -- Nortel
  -- 3Com
  -- Avaya
  -- Mitel
  -- Siemens
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 %)
Alcatel-Lucent SIP Support

- SIP-call Security: **No TLS** – Transport Layer Security (IPsec for call control), some secure RTP (to SIP applns server), authentication
- Extent of validated SIP interoperability:
  - 3rd-party SIP phones: **3 vendors**
  - Carrier services via SIP trunks: **18 (based on IETF, SIP Forum and TISPAN specs)**
  - Applns server works with: **2 other vendors’ SIP call controllers**
Avaya SIP Support

• Main SIP-supporting platform(s): SIP Enablement Services (SES), a separate server from Comm Mgr

• Is SIP primary call control? Only via separate SES. Primary is proprietary H.323. H.248 too, vendor says

• Vendor offers SIP phones? Yes (half-doz models + soft)

• SIP standard RFC features: 5 (10%)

• SIP draft-based features: 0 (0 %)

• SIP proprietary headers or features codes: 55 (90 %)
Avaya SIP Support

• SIP-call Security: TLS, no secure RTP, authentication
• Extent of validated SIP interoperability:
  – 3rd-party SIP phones and gateways: 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 models + 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 appln 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): Vendor’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/
There's no end to better.