SIP Trunk Interworking: How the SIP Forum is Improving Interoperability Between SIP-PBXs and Service Provider Networks

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Agenda

• SIPconnect1.1 Recommendation (David Hancock)

• SIPconnect1.1 Interoperability Program (John Berg)

• SIPconnect1.1 Compliance Program (James Swan)

• Q & A
SIPconnect1.1 Recommendation
Problem: SIP Trunk Interworking Issues

SIP Trunk products don’t conform to a single SIP standard.

SIP Trunk

SIP-PBX-1

SIP-PBX-2

SIP-PBX-3

\ldots

SIP-PBX-n

\begin{itemize}
  \item Service Provider–A
  \item Service Provider–B
  \item Service Provider–C
  \item \ldots
  \item Service Provider–X
  \item Service Provider–Y
  \item Service Provider–Z
\end{itemize}

Time/resources required to "qualify" each new PBX / SP-network combination.
Solution: SIPconnect1.1

Defines a common SIP profile for SIP-PBXs and SP-SSEs
Guiding Principles

• Keep it simple
  – Support only basic SIP Trunk capabilities & features
  – Resolve the most common interworking issues
  – Minimize list of required SIP extensions

• Level of detail

  Less Detailed
  • More vendor innovation
  • Faster time-to-market

  Level of detail

  More Detailed
  • Improved interworking

• SIPconnect1.1 errs on the side of less-detail
• Future SC1.1 updates may nudge needle to the right
Documents MUST requirements for SIP-PBX, SP-SSE, and Media Endpoints

- Registration mode (GIN)
- Static Mode
- SIP Digest / TLS
- Basic Call Features
- Direct Inward / Outward Dialing
- Calling name/number delivery
- Call Forwarding
- Call Hold
- Call Transfer
- Emergency Calls
- FAX & DTMF relay
SC1.1 Mandatory Standards

• **SIP/SDP**
  – IETF RFC 3261 SIP: Session Initiation Protocol
  – IETF RFC 3264 Offer/Answer
  – IETF RFC 3323 Privacy header
  – IETF RFC 3325 P-Asserted-Identity header
  – IETF RFC 3327 Path header
  – IETF RFC 4566 SDP: Session Description Protocol
  – IETF RFC 5876 Updates to Asserted Identity
    – IETF RFC 6140 Registration for Multiple Phone Numbers in SIP (GIN)

• **Media**
  – IETF RFC 3550 RTP
  – IETF RFC 3389 Comfort Noise
  – IETF RFC 4733 RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
  – ITU-T T.38 Real-time Group-3 FAX over IP networks

• **TLS**
  – IETF RFC 5280 X.509 Public Key Infrastructure Certificate and Cert Revocation List Profile

New registration extension
Adoption of SIPconnect1.1

• Support in today’s products (anecdotal)
  – Majority of procedures are commonly supported
    • It’s mostly a configuration problem
  – Some procedures less widely supported
    • E.g., RFC 6140 (aka "GIN" Registration)

• CableLabs activities
  – Developed a SIPconnect1.1-compliant Reference Implementation based on Asterisk
  – Demonstrated "GIN" registration interworking between RI and two different Soft-Switch vendor implementations
  – Submitted RFC6140 patch to Asterisk open-source

• 3GPP activities
  – CT1 is adding support for RFC 6140 to IMS Rel-11
SIPconnect1.1 Enhancements

- Minimize changes to encourage adoption
  - Correct errors/ambiguities
  - Avoid scope increase

- Some updates already in pipeline
  - Enhance emergency procedures to align with industry standards
  - Correct registration issue; treat REGISTER for unknown AOR same as normal REGISTER

- Tech W-G may make additional updates (hopefully minor)
  - Resolve interworking issues discovered during interop testing
  - Fix errors/ambiguities reported by product developers

- Release date TBD
A quick look at "GIN" Registration
A non-SC1.1 Registration Variant

[1] REGISTER
To: sip:+13035551212@sp.com
Contact: sip:192.168.10.20

SIP-PBX
AOR: sip:+13035551212@sp.com
Contact: sip:192.168.10.20

SP-SSE
Location Database
AOR
sip:+13035551212@sp.com
sip:+13035551213@sp.com
sip:+13035551214@sp.com
Location
sip:192.168.10.20

[2] INVITE sip:+13035551212@sp.com
[3] INVITE sip:+13035551213@sp.com
Route: sip:192.168.10.20

Potential routing problem: ambiguity over who "owns" sp.com
RFC 6140 (aka "GIN") Registration

[1] REGISTER
    Require: gin
    To: sip:pbx-1@sp.com
    Contact: sip:192.168.10.20; bnc

SIP-PBX

AOR: sip:pbx-1@sp.com
Contact: sip:192.168.10.20

SP-SSE

Location Database

<table>
<thead>
<tr>
<th>AOR</th>
<th>Location</th>
</tr>
</thead>
<tbody>
<tr>
<td>sip:<a href="mailto:pbx-1@sp.com">pbx-1@sp.com</a></td>
<td>sip:192.168.10.20</td>
</tr>
<tr>
<td>sip:<a href="mailto:+13035551212@sp.com">+13035551212@sp.com</a></td>
<td>sip:+13035551212@192.168.10.20</td>
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</tr>
</tbody>
</table>

[2] INVITE sip:+13035551213@sp.com

[3] INVITE sip:+13035551213@192.168.20.20

Request routing per normal RFC3261 procedures
Interoperability Test Program
SIPconnect-IT Task Group

• Goal
  – Accelerate industry adoption of SIPconnect1.1

• Co-chairs
  – Alan Johnston – Avaya
  – Robert Kinder - Cox

• Deliverables
  – SIPconnect1.1 Interoperability Test Program
  – SIPconnect1.1 Compliance Test Program
Scope of Interoperability Testing

• Evaluate ability of multi-vendor products to interoperate
• Tests based on requirements; looser approach
• Provide data on
  – Product readiness
  – Feedback to Technical Working Group
  – Test plan enhancements
Interop/Compliance Test Comparison

**Interoperability Testing**
- Focus on Interoperability
- Some degree of SC1.1 compliance
- Multi vendor testing
- No third party test tools
- Outcomes based on ‘what works’
- Includes tests for interworking issues

**Compliance Testing**
- Focus on requirements
- Full conformance testing
- Single product testing
- Test tools form the basis of testing
- Outcomes based on compliance
- SC1.1 MUST requirements only
Interop Test Plan Organization

SC1.1 Interop Test Plan

- Requirements Testing
- Interworking Testing

- Registration Mode
- Static Mode
- Security
- Basic Calls (DOD/DID)
- Business Voice Features
- Session & Media Interaction

Use Cases
- Early Media – DOD Call
- Call Forwarding Redirects
- Additional tests to be added
Registration & Static Modes

• Registration Mode
  – RFC 6140 (GIN) registration
  – Registration failures, retries and failover

• Static Mode
  – Statically configured routes from SP-SSE to SIP-PBX

• Common requirements
  – DNS
  – TCP
  – SIP Digest
  – TLS
Basic Calls /Business Voice Features

- Originating calls from the SIP-PBX (DOD)
- Originating calls from the SP-SSE (DID)
- INVITE messages are correctly formed
  - Request URI, PAI header, From header, etc.
- Business Voice features as defined in SIPconnect1.1:
  - Calling Name and Number Delivery
    - Ability to enable/disable privacy by the SIP-PBX and the SP-SSE
  - Call Forwarding
  - Call Hold
    - By both the SIP-PBX and the SP-SSE
  - Call Transfer
    - Includes Blind, Attended and Early Media scenarios
  - Emergency Services
Media and Session Interactions

• SDP (RFC 4566)
• Offer/Answer Model (RFC 3264)
• Codec negotiation
  – Presently G.711 and G.729
• DTMF and telephone events (RFC 4733)
• T.38 fax
Interworking Issues

• Based on use cases derived from interoperability testing and observed issues in field deployments
• Replicate interworking issues between SIP-PBX and SP-SSE in a lab environment
• May point to further clarifications required in the spec
• Purpose is to identify potential interoperability issues between vendor products
Requirements Coverage

• 169 Total SC1.1 MUST requirements (SIP-PBX/SP-SSE)
  – Includes requirements from RFC 6140 for GIN registration
• Test Plan covers:
  100% of testable MUST requirements
  89% of all MUST requirements

Note: 19 requirements excluded from interop testing will be re-evaluated for compliance testing.
Next Steps

• Select venue for first interop
  • Tentatively scheduled for October

• Enhance interworking issues coverage
  • Current draft test plan available here

• Seeking participation in bi-weekly calls
  – Contribute to the scope of interops
  – Participate in developing compliance test program

• Participation will make this program successful
Compliance Test Program
Compliance Test Program Highlights

• Overall goal
  – Test product conformance with SIPconnect1.1 MUST requirements

• Deliverables
  – Compliance Test Plan (leverage Interop Test Plan)
  – Compliance Test Platform (tool-based)
  – Set of test scripts to test SP-SSE and SIP-PBX compliance with SC1.1
  – SIPconnect1.1 compliance branding program

• Start work Q3 of 2012

• Target program-ready date – Q1 2013
Compliance Test Plan

- SC1.1 Interop Test Plan
  - Requirements Testing
  - Interworking Testing

- SC1.1 Compliance Test Plan
  - Requirements Testing
  - Registration Mode
  - Static Mode
  - Security
  - Basic Calls (DOD/DID)
  - Business Voice Features
  - Session and Media Interaction

- Expand requirement coverage and add detail
- Serve as input to script developers

Compliance Test Scripts
SP-SSE test scripts emulate a SC1.1-complaint SIP-PBX to verify that Service Provider Network supports SIPconnect1.1 MUST requirements.

A separate set of scripts emulating the SP-SSE would be required to validate compliance of SIP-PBX.
Compliance Test Tools

• Several test tools are being evaluated
  – Should be automated/scriptable
  – Should be free / open-source

• CableLabs has contributed two test tools
  – SIPconnect1.1 Reference Implementation
    • Based on Asterisk
  – PCsim2
    • Scriptable test platform that emulates SIP call flows
  – Both tools capable of emulating SIP-PBX and SP-SSE
  – Tools will be available via SIP Forum web site
**Interop/Compliance Program Timeline**

**2012**
- **Q3**
  - Draft Test Plan available
  - Start work in July
- **Q4**
  - Interoperability Test Program
  - Target first interop event week of Oct 8
- **2013**
  - Q1
    - Use draft Interop Test Plan as base
    - New interworking issues may drive future SC1.1 spec work
  - Q2
    - Future Interops (TBD)
    - Target program-ready date (TBD)
Open Issues

• What exactly do we certify?
  – E.g., when SP-SSE is comprised of multi-vendor products
• How does branding address optional functionality?
  – E.g., Static vs. Registration mode
• Certification program lifecycles?
  • Compliance Test Tool TBD
  • Compliance Program launch date TBD
Questions/Comments?
Contact Us

• SC-IT mailing list
  – sipconnectit@sipforum.org

• David Hancock
  – D.Hancock@cablelabs.com

• John Berg
  – j.berg@cablelabs.com

• James Swan
  – jmswan@iol.unh.edu
Online Resources

• SC-IT Task Group Charter
  – http://www.sipforum.org/content/view/393/285/

• Mailing List
  – http://www.sipforum.org/mailman/listinfo/sipconnectit

• Document repository
  – http://www.sipforum.org/component/option,com_docman/task,cat_view/gid,120/Itemid,261/

• Draft Interoperability test specification
  – http://www.sipforum.org/component/option,com_docman/task,cat_view/gid,125/Itemid,261/