

SIP Trunk Interworking:

**How the SIP Forum is Improving Interoperability Between
SIP-PBXs and Service Provider Networks**

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John Berg – CableLabs

James Swan – University of New Hampshire



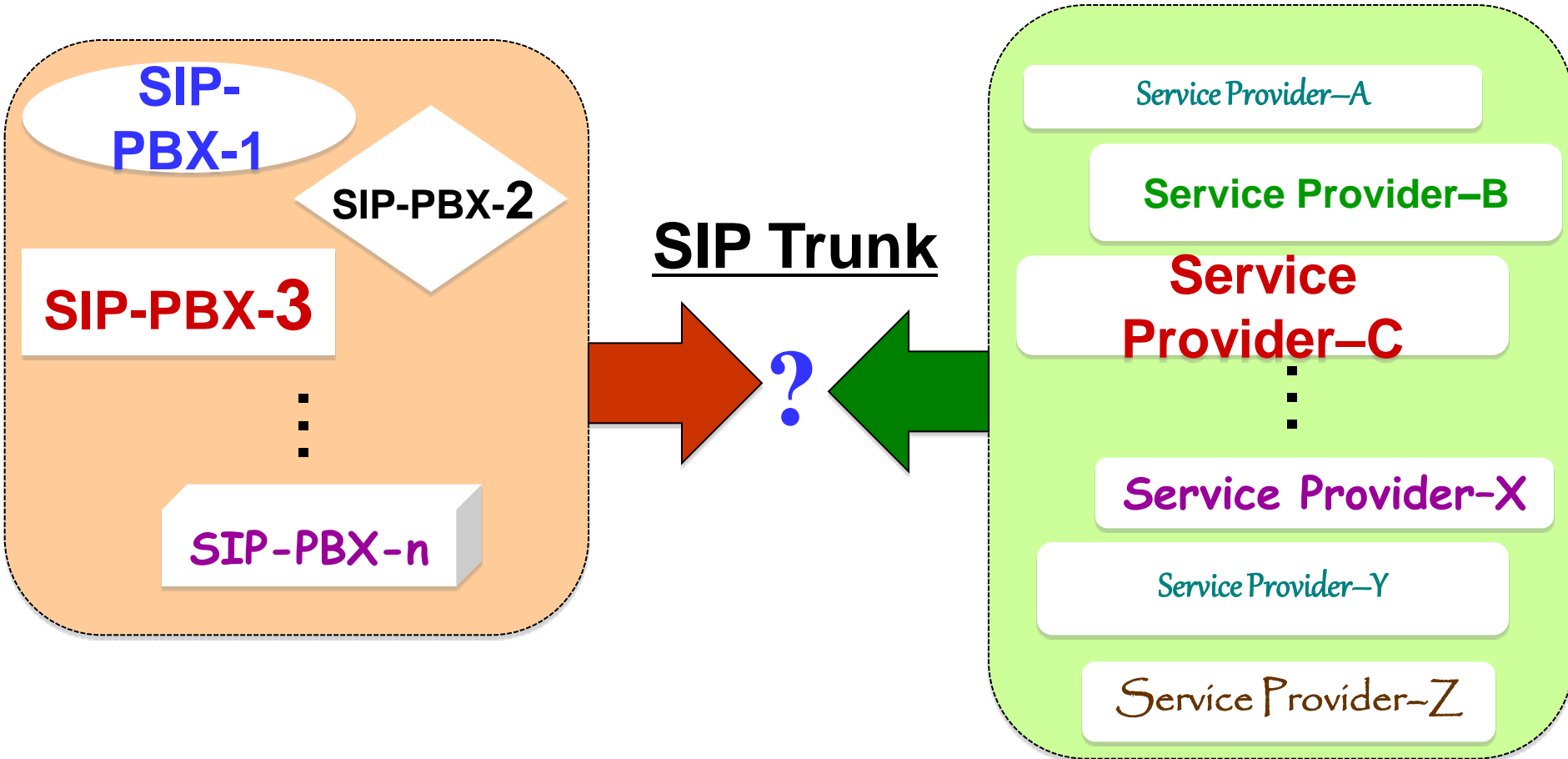
Agenda

- 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

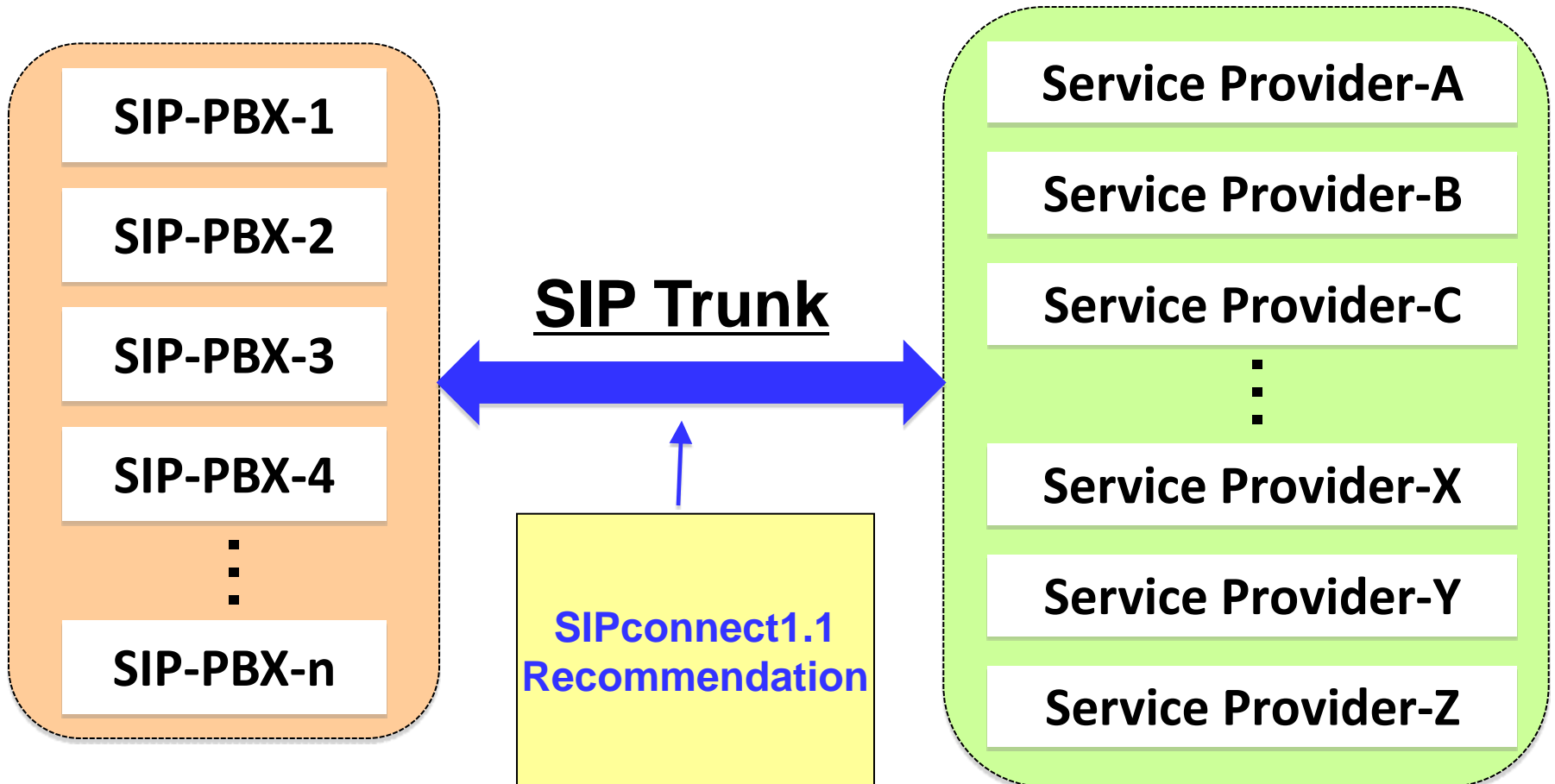
SIP Trunk products don't conform to a single SIP standard



Time/resources required to "qualify" each new PBX / SP-network combination

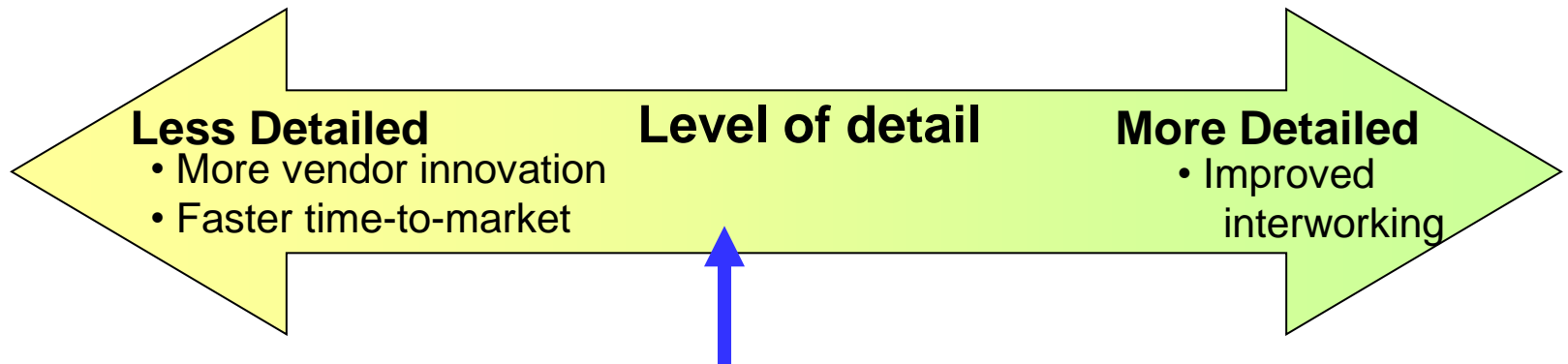
Solution: SIPconnect1.1

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



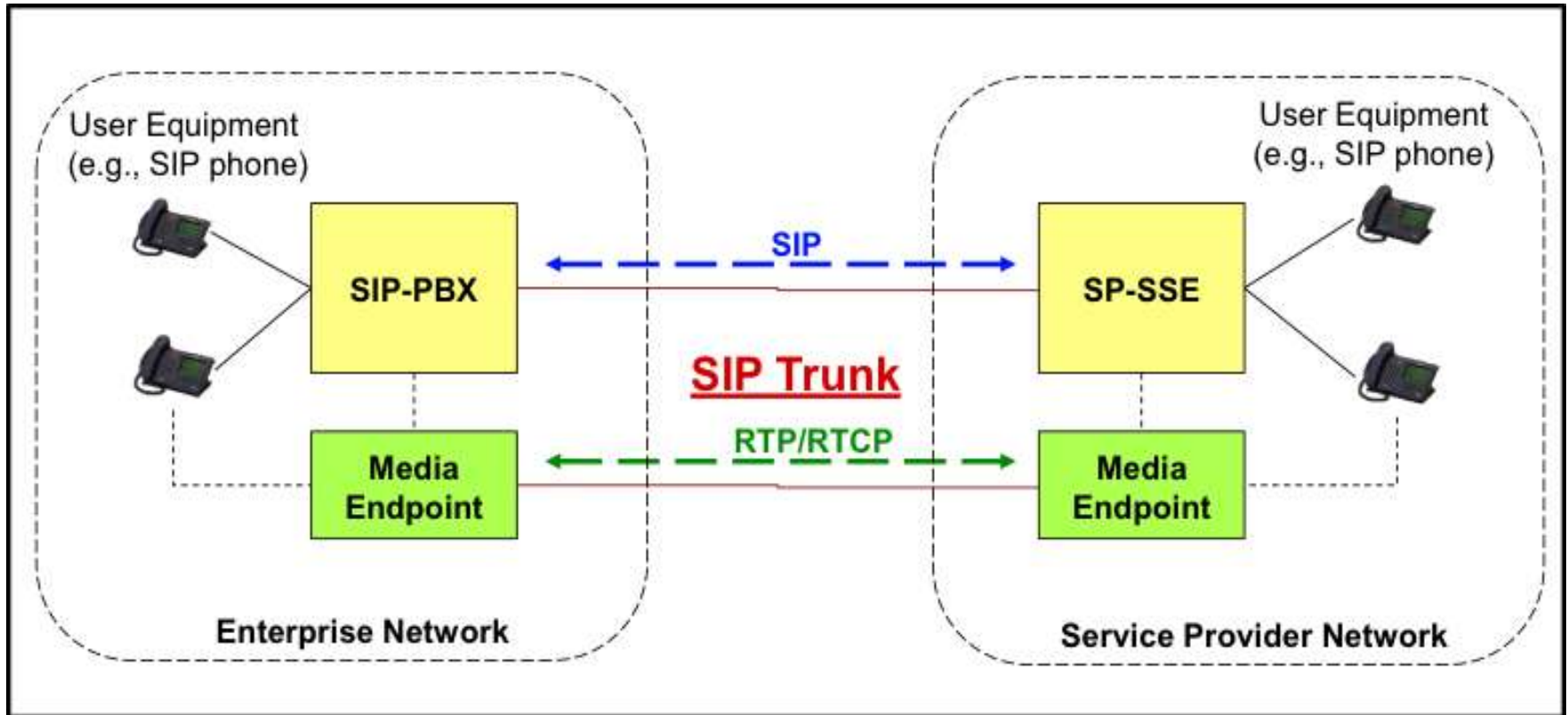
Guiding Principles

- 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



- SIPconnect1.1 errs on the side of less-detail
- Future SC1.1 updates may nudge needle to the right

SIPconnect1.1 Recommendation

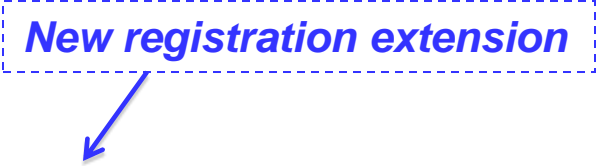


Documents MUST requirements for SIP-PBX, SP-SSE, and Media Endpoints

- 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

AOR

Location

sip:pbx-1@sp.com	→ sip:192.168.10.20
sip:+13035551212@sp.com	→ sip:+13035551212@192.168.10.20
sip:+13035551213@sp.com	→ sip:+13035551213@192.168.10.20
sip:+13035551214@sp.com	→ sip:+13035551214@192.168.10.20

[3] INVITE sip:+13035551213@192.168.20.20

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

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

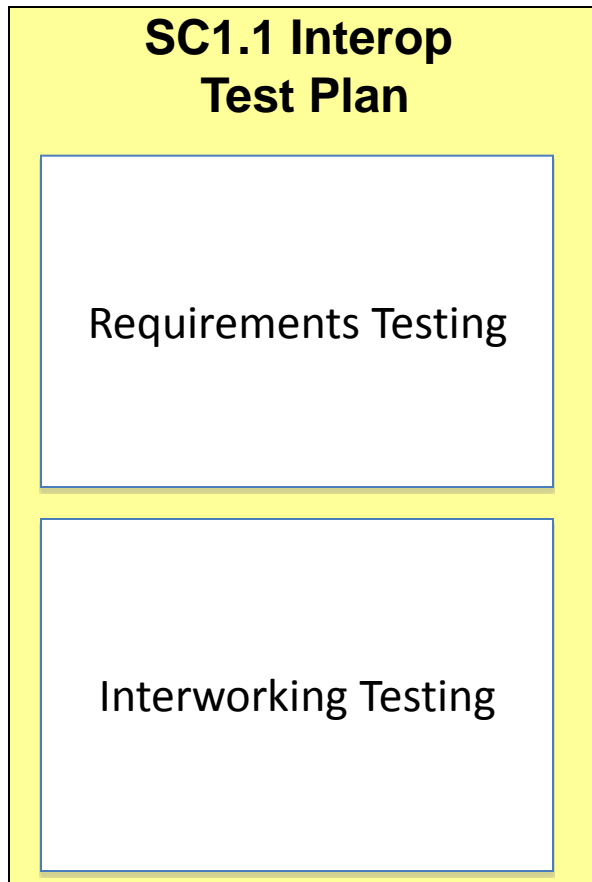
Interoperability Testing

- 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



- 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

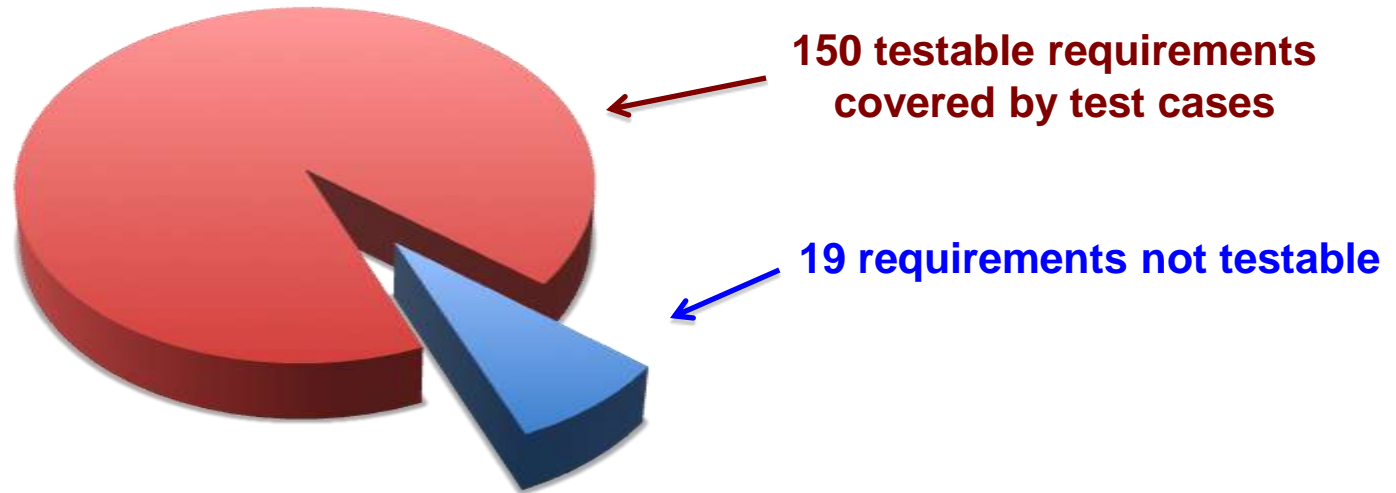
Media and Session Interactions

- 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

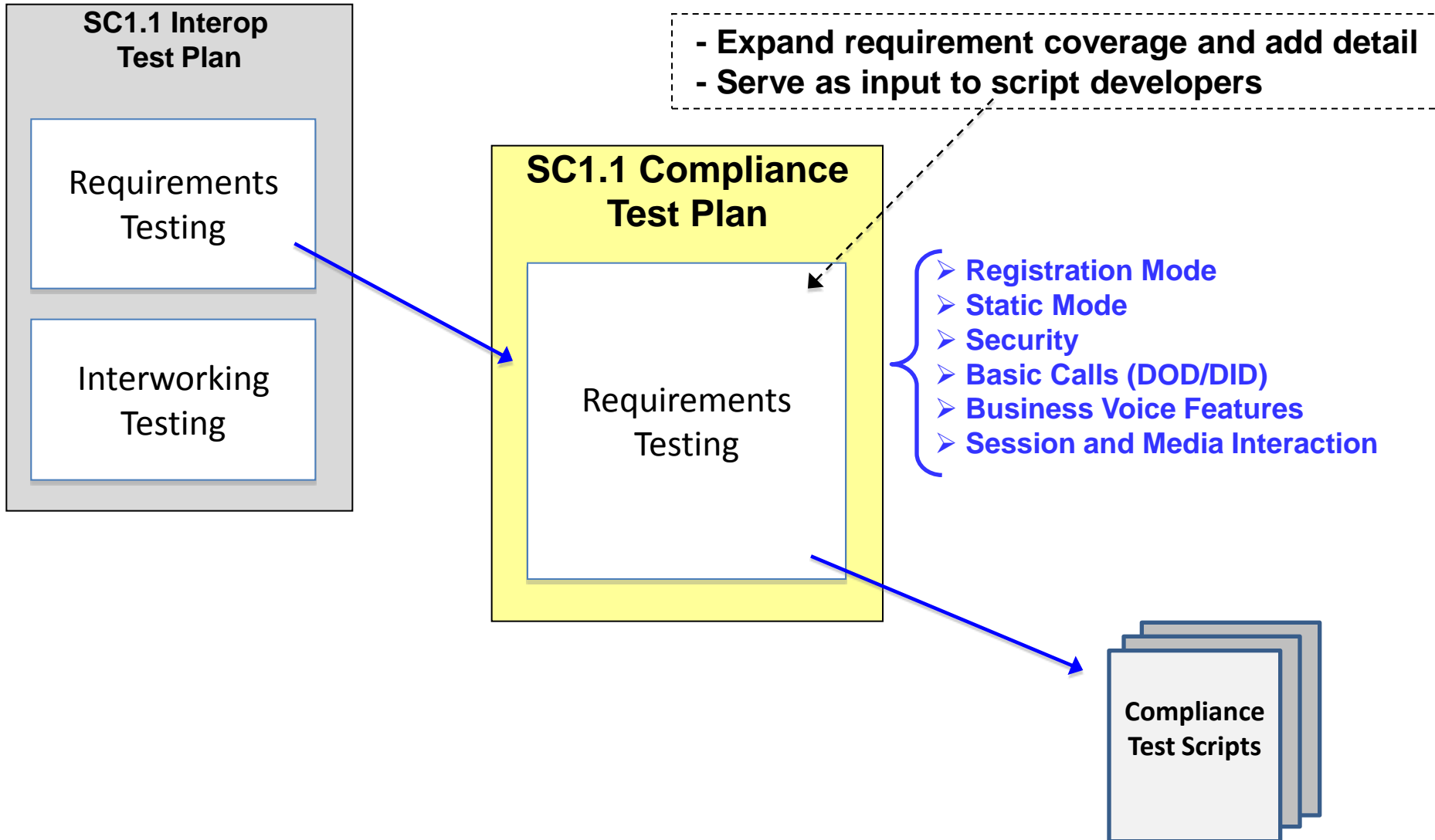
- Select venue for first interop
 - Tentatively scheduled for October
- Enhance interworking issues coverage
 - Current draft test plan available [here](#)
- Seeking participation in bi-weekly calls
 - Contribute to the scope of interops
 - Participate in developing compliance test program
- Participation will make this program successful

Compliance Test Program

Compliance Test Program Highlights

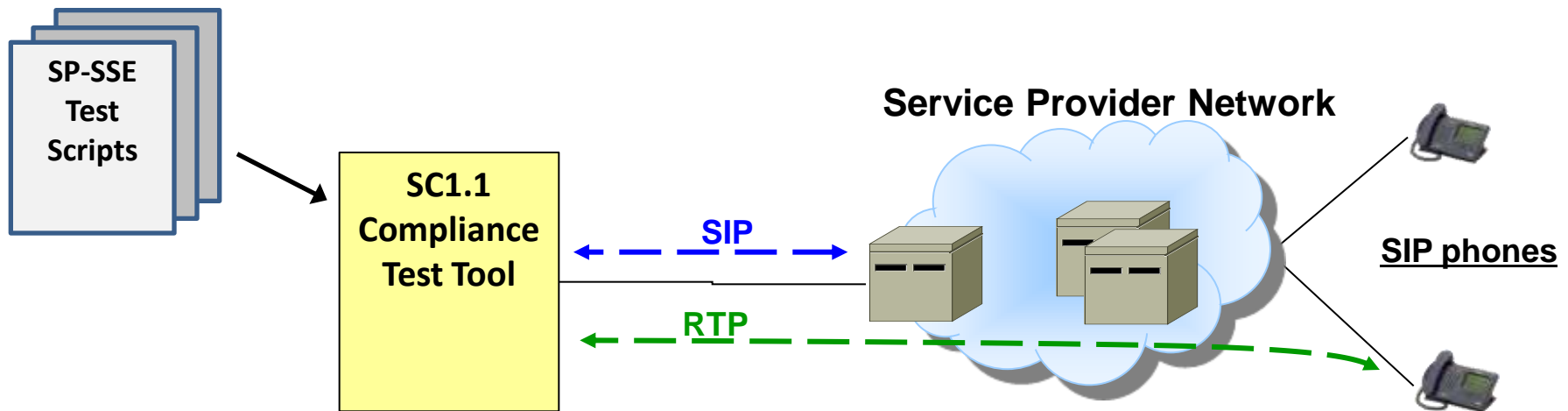
- Overall goal
 - Test product conformance with SIPconnect1.1 MUST requirements
- Deliverables
 - Compliance Test Plan (leverage Interop Test Plan)
 - Compliance Test Platform (tool-based)
 - Set of test scripts to test SP-SSE and SIP-PBX compliance with SC1.1
 - SIPconnect1.1 compliance branding program
- Start work Q3 of 2012
- Target program-ready date – Q1 2013

Compliance Test Plan



Compliance Test Platform

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

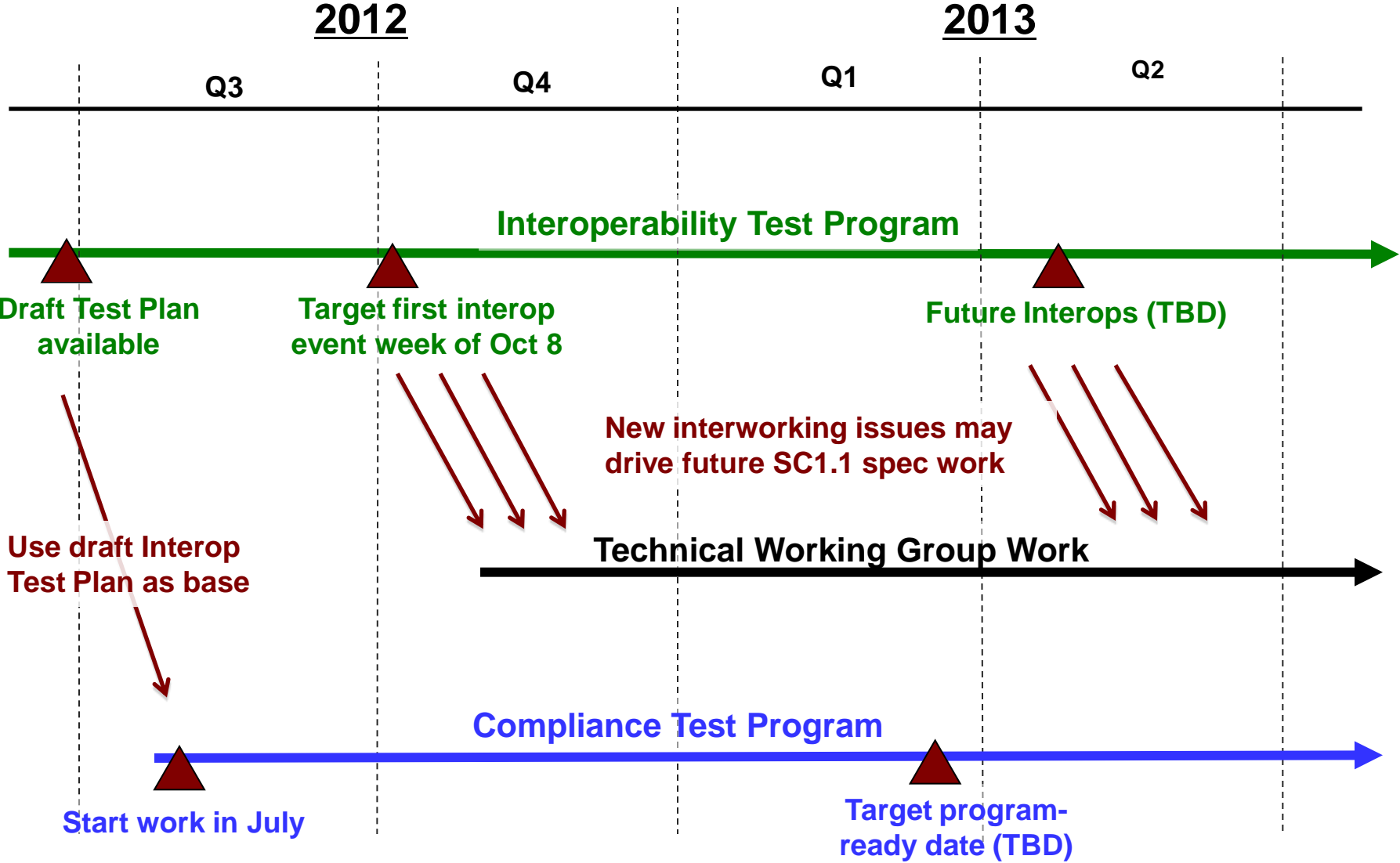


- A separate set of scripts emulating the SP-SSE would be required to validate compliance of SIP-PBX

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



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
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- John Berg
 - j.berg@cablelabs.com
- James Swan
 - jmswan@iol.unh.edu

The logo for SIPFORUM, with "SIP" in blue and "FORUM" in green, and a green circular arrow icon between the two words.The logo for CableLabs, with "CableLabs" in blue and "REVOLUTIONIZING CABLE TECHNOLOGY" in smaller blue text below it.

Online Resources

- SC-IT Task Group Charter
 - <http://www.sipforum.org/content/view/393/285/>
- Mailing List
 - <http://www.sipforum.org/mailman/listinfo/sipconnect>
- 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/