

The following list of topics and issues is being submitted for consideration as part of scope for the SIPconnect 1.1. These suggestions are based on our practical experience as a leading contributor to the SIPconnect Technical Recommendation in the past and our real world deployment experience since 2005.

**Document Organization.** We propose a new format for the document that supports the SIPconnect Compliant certification process. The reference architecture should be revisited to ensure it captures all key product types (i.e. PBX, service provider, proxy, etc.) in the end-to-end SIPconnect architecture. Each requirement in the document should be clear as to which elements of the architecture it applies. This should be done first in the process so all sections of the document reflect terminology that is consistent with the architecture and its labels.

**Update the RFC Table.** The RFC table should be updated to include new RFCs that apply and remove ones that are no longer appropriate.

**Call Transfer / Hairpin scenarios.** We need to address an agreed upon approach to call flows that were not covered in SIPconnect 1.0, including:

- Call Forward Off-Net with preservation of original calling number and optional release of media
- Call Transfer Off-Net (blind and supervised)

There are also more complex issues related to how PBX performs call handling that is not visible to the SP. Consider a scenario where PSTN A calls Enterprise User B who transfers the call to Enterprise User C and Enterprise User C then transfers the call to PSTN D. The AS has one session established with B and then another session established with C. Then C wishes to REFER (w/ Replaces) the call to the A-B session. The AS must allow one user to REFER sessions to a session owned by another user.

**Multi-Location Scenarios.** Many customers want to deploy a single PBX that serves multiple business locations. Cbeyond supports this capability today using REINVITE to allow a PBX to establish a call to our network and then re-invite a phone to anchor media directly to our network. We should discuss multi-location support using a single PBX with SIP Trunking. A standard approach to providing this support should be considered in 1.1.

**TLS.** Our observation from the field is that TLS, while mandated in SIPconnect 1.0, is not widely deployed. The obstacles to its adoption seem large, especially given the proliferation of enterprise SIP ALGs and firewalls that require visibility of the signaling path to address NAT and security concerns. We should make TLS optional and find a more palatable security solution to require that more people are willing to adopt.

**Parent/Child Concept from Early SIPconnect Drafts.** Most SIPconnect deployments in production rely on a PBX to REGISTER to determine the IP address of the PBX and authenticate it via DIGEST during the registration and subsequent INVITES. Moreover, many service providers employ a parent/child framework and update the contact information for multiple or all DID/Users on a PBX using the single registration of a parent user of the system. All associated users also share the same authentication credentials. This approach has proven very practical in SIPconnect 1.0 even though it is not explicitly called out in the SIPconnect 1.0 recommendation. This issue needs to be revisited. We need to understand

the merits of this approach and either ratify it in 1.1 or find another approach that solves the challenges this solution addresses.

**Provisioning.** Various SIPconnect products and services already support a basic provisioning framework wherein a PBX consumes XML data provided by a service provider that includes information needed for automatic configuration such as DIDs available on the service, outbound SIP proxy URL, SIP username and password, etc. Beyond would like to see a standardized schema and post/retrieval method agreed upon.

**Media Requirements.** We need to revisit and refine the media requirements to address at least the following questions, and probably more.

- What CODECS should be required?
- Should T.38 now be required exclusively for FAX
- Should DTMF Relay now be exclusively required?
- Should we drop the requirement for ulaw to alaw transcoding?

**Video and Other CODECS.** We should add an explicit statement that SIPconnect providers should not block video and other optional CODECS when selected by two customers who are directly connected via SIPconnect.

**Trust Domain.** The idea that the PBX is part of the SP's trust domain should be removed. The use of the P-Preferred header can convey the identity of the user originating the request (of course, it has to be authenticated).

**Conveying originating identity on PBX to SP calls.** The P-Preferred header should be used on requests to indicate the identity of the caller (from a billing and feature application standpoint) while reserving the FROM header to convey the desired calling number. This solves the issue of "Caller ID Preservation" where the PBX bridges the call in a Find-Me/Follow-me feature and sends a request back to the service provider. It also allows the PBX to indicate some "off-net" number (Caller ID Spoofing) as the calling number (such as their main Toll-Free number).

**SBC Integration / Routing SP to PBX requests through an SBC** This is addressed pretty well in the TISPAN Business Trunking document. It based on loose routing as specified in 3261. The SP stores the SBC as the "route" and the registered Contact Address for the trunkgroup. When the SP needs to route the call to the PBX, it adds a Route header to the request containing the route and the registered contact address. This is "SBC friendly" in that the SBC can identify the registered PBX based on the Contact identity in the Route header (see section 6.1.5.2 in the attached TISPAN doc DTS/TISPAN-02042-NGN-R2).