**Contribution**

**TITLE: Comments on the IP Interconnect Specification**

**SOURCE\*:** Alcatel-Lucent

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**ABSTRACT**

This document proposes a number of changes to the IP Interconnect specification.

* *The use of 100rel procedures are required across the NNI. To avoid confusion, the example is modified to use an optional extension – preconditions.*

## 5.1 Extension Negotiation

SIP entities involved in session peering SHOULD be configured in such a way that they do not require any SIP extensions, beyond those mandated by this document, to be supported by the peer Carrier (SIP Service Provider) network. When sending an out-of-dialog request to a peer Carrier network, SIP entities involved in session peering SHOULD include a Supported header field identifying all the extensions supported by the sending network.

SIP entities involved in session peering MAY support configuration controls to disable certain extensions based on bilateral agreement between peer Carrier networks. For example, a SIP entity involved in session peering could be configured to remove ‘preconditions’ from the Supported header in order to disable the use of reliable provisional response (PRACK).

Note: Policies that limit or block the use of SIP extensions should be applied with care, since their application tends to disable SIP's native extension negotiation mechanism, and therefore inhibit the deployment of new services.

When sending a dialog-initiating request to a peer Carrier network, SIP entities involved in session peering MUST identify all supported SIP requests in the Allow header field

* *Add a reference to the IETF Overload Control document (still in I-D form).*

### 5.4.2 Congestion Control

Carrier's MUST support SIP Overload Control with mandatory support of the default algorithm [draft-ietf-soc-overload-control-15]. Carrier's MAY optional support the Rate Based algorithm based on bilateral agreement between two carriers.

A Carrier network MAY impose limits on the number of simultaneous calls, and the incoming rate at which it will accept calls, from a peer. On receiving a dialog-initiating request that exceeds such limits, the receiving Carrier network MUST respond with a 503 (Service Unavailable) response. A Carrier network receiving a dialog-initiating request MUST limit the use of the 503 responses to reporting congestion at the ingress point. A terminating Carrier network MUST NOT send a 503 response to an originating Carrier network to report congestion or other failures that are internal to the terminating Carrier network. For example, a 503 response generated by a SIP signaling entity within a terminating Carrier network should be consumed within the terminating network, and should not be propagated across the peering interface to the originating Carrier network (i.e., avoid sending a 503 response to an originating peer if the same failure is likely to be encountered when the call is retried via an alternate route).

On receiving a 503 (Service Unavailable) response from a peer Carrier network, the receiving Carrier network MUST limit the scope of the response to the call on which it was received (i.e., a 503 response has no affect on the routing of subsequent calls to the peer). Also, the receiving Carrier network MUST attempt to consume the 503 response from a peer as close to the egress signaling point as possible, and avoid propagating the response back toward the originating CMS or E-DVA. Specifically, on receiving a 503 response to a dialog-initiating request that was sent to a peer Carrier network, the receiving Carrier network MUST:

1. terminate the current transaction,
2. ignore the Retry-After header field if one is present, and
3. attempt to route the call via an alternate peering interface (i.e., do not attempt to route the call via the same peering interface since it may encounter and aggravate the same overload condition).

* *A-law is not a typical North American codec. It should not be marked as mandatory support.*

### 5.5.2 Codecs

Narrow Band codecs reproduce the audio bandwidth of the PSTN. The following codecs, widely used in IP based voice networks, shall be supported as described in the tables below. Codecs in the Group 1 column in each table MUST be supported for both transmission and reception across the NNI. Codecs in the Group 2 columns in each table SHOULD be supported for both transmission and reception across the NNI.

|  |  |
| --- | --- |
| **Group 1. Mandatory Narrow Band codecs** | **Group 2. Optional** |
| G.711 μ-law 64 kbit/s | G.711 A-law |
|  | G.723.1 (quality impairments have to be considered using this codec) |
| G.729, G.729a, G.729b, G.729ab 8kbit/s | G.726 |
|  | AMR-NB |

Table 1 – Mandatory and Optional Narrow Band Codecs

When wide band audio is being used, the following wide band codecs, widely used in IP based voice networks, shall be supported as described in the tables below. Codecs in the Group 1 column in each table MUST be supported for both transmission and reception across the NNI. Codecs in the Group 2 columns in each table SHOULD be supported for both transmission and reception across the NNI.

|  |  |
| --- | --- |
| **Group 1. Mandatory Wideband codecs (\*)** | **Group 2. Optional Wideband codecs** |
| G.722 (generally used by fixed network operators) |  |
| G.722.2 (AMR-WB, generally used by mobile network operators) |  |

**Table 2 – Mandatory and Optional Wideband Codecs**

* *Transcoding should be minimized. If the originating network does not support wide band codecs, then wide band codecs should not be prioritized over the highest priority narrowband codec offered by the originating network.*
* *Statement regarding routing across country borders is not necessary.*

**5.5.4 General guidelines**

The following general guidelines aim to provide default rules for codec choice and transcoding responsibility:

1. Transcoding should generally avoided;
2. If the originating network supports both wideband codec, then if both narrowband and wideband codecs are offered in a VoIP session the wideband codecs should be placed in top priority. Otherwise, the highest priority narrowband codec offered by the originating network shall be placed in top priority.
3. Wideband codec continuity offers the optimal quality; Service Providers should offer a fallback to narrowband codec that is universally supported (e.g. G.711) along with its supported high quality codec(s).
4. Transcoding to narrowband codecs must be avoided unless it is the only way for a call to be successfully established;
5. the order of codec/packetisation period preference is determined by the originating terminal and should be honoured wherever possible;
6. if the call is to be routed to a TDM network, only one transcoding is recommended. If required, it should be performed during the voice over IP/TDM conversion;in case no common codec can be used between both end Service Providers, in the first instance it is the responsibility of Service Providers to support transcoding in order to ensure successful voice interoperability for their services

* *Reliable provisional response (100rel) procedures are required across the NNI. The offer/answer procedures should take this into consideration.*

## 6.1 Basic Call Setup

This section describes the procedures at the peering interface required to establish a 2-way session for a basic voice call. In this case it is assumed that no originating or terminating features are applied (no call blocking, forwarding, etc), and that the called line is available to accept the call. Also, this section describes the session establishment procedures when the call is initiated by the originating SIP User Agent itself, and not via a 3rd party in support of features like click-to-call. Two-way call establishment using 3rd Party Call Control (3PCC) procedures is covered in Section .

SIP entities involved in session peering MUST support the SDP offer/answer procedures specified in with the consideration that reliable provisional responses MUST be used as specified in [RFC 3262]. The originating Carrier network SHOULD include an SDP offer in the initial INVITE. The terminating Carrier network MUST include an SDP answer reliable response to an INVITE received with an SDP offer. The terminating Carrier MUST include an SDP offer in the first reliable response to an INVITE received without an SDP offer. Once an SDP answer has been provided in a reliable response, it MUST not be repeated in subsequent responses. For example, if an SDP answer if provided in a reliable 183 Session Progress response, it must not be repeated in the 200 OK (INVITE).

The terminating Carrier network MAY also include an SDP body in a provisional 18x response or reliable response (e.g., PRACK).

**Note**: If the provisional and final responses are on different dialogs (say, when the INVITE is forked), the SDPs may be different between the various responses.

SIP entities involved in session peering that advertise support for different but overlapping sets of codecs in the SDP offer/answer exchange for a given call MUST negotiate a single common codec for the call. An SDP answer MUST contain only a single codec (plus additional auxiliary codecs such as DTMF), per media stream, selected from the offered set of codecs.

* *Since reliable provisional responses are required, the use of forking is not necessary. Request-Disposition header should be honored when offered to prevent returning forked responses not supported by upstream node.*

## 6.4 Forking the INVITE

For each terminating media endpoint that requires an early media session to be established with the originating line, the terminating Carrier network MUST signal the attributes of the terminating media endpoint to the originating Carrier network within the SDP of a 183 (Progressing) response.

If terminating Carrier needs to modify the SDP, the Carrier SHOULD offer the modified SDP in an UPDATE request.

Alternatively, with bi-lateral agreement, the terminating Carrier network MAY utilize forked responses to ensure that 18x/200 responses containing different SDP copies are not sent within the same dialog. This MUST only be used if it had not previously received a Request-Disposition header [RFC 3841] preventing the use of forking, (e.g., Request-Disposition: no-fork). The terminating Carrier network does this by specifying a different tag parameter in the To header field for each provisional response that contains a unique SDP, as if the INVITE had been sequentially forked.

* *Since both the History-Info header and the Diversion header may both be present, a precedence statement is needed for the receiving side.*

## 6.9 Call Forwarding

Carrier's MUST support the History-Info Header and SHOULD support of the SIP Diversion header for a period of time in order facilitate interoperability. When both headers are sent, the sender MUST ensure that they are semantically identical.

If the History-Info header and the Diversion header are both received by a carrier supporting both headers, the History-Info header MUST take precedence (i.e., the Diversion header will be ignored).

If a Carrier offers call-forwarding services to its users, then the forwarding Carrier network MAY remain in the signaling path of the forwarded call in order to support separate billing for forward-from and forward-to legs. A Carrier network that is required to remain in the signaling path of a forwarded call based on local policy MUST do so using one of the following procedures:

1. forward the INVITE to the forward-to-user while remaining in the signaling path as a SIP Proxy or B2BUA, or

2. respond to the initial INVITE with a 302 (Moved Temporarily) response with a Contact header field containing a private URI that points back to the forwarding Carrier network.

* *Overlap signaling is not applicable for North American networks.*

#### 7.1.2.2 Modes of signalling

Overlap signaling is not applicable for North American networks.