

RFC for Proposed practices for interconnecting SIP providers.

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", AND "OPTIONAL" in this document are to be interpreted as described in BCP 14, IETF RFC 2119 [1].

- 1) What forms of network interconnect will Service Providers use to exchange calls?

(i.e., SPs will exchange calls over the Internet, via contracted private peering connections, and/or via voice peering exchanges)

- 2) Can we assume that Interconnects will be SIP over IP?
- 3) What are the Quality, Reliability, Availability, etc.,
A requirement of inter-SP interconnects?

(Number of 'nines' available/acceptable seconds of outage, packet Loss, latency, jitter, etc.)

- 4) Should Service Providers rely on 'dynamic' protocols such as ENUM for discovery of inter-SP telephony routing information?
- 5) What are addressing requirements for Interconnect?
(From both network/IP and URI/Phone Number perspectives)
- 6) What SIP elements and standards must be supported for inter-SP Calls?
- 7) Are there any network security requirements, beyond 'normal' Practices, that are Specific to inter-SP SIP Interconnects?
- 8) Should support of specific codec's be mandated? (i.e., G.711 A-law and U-law, G.729a)
- 9) What levels and forms of Service Level / Quality of Service Monitoring and/or reporting should be required for inter-SP Interconnects?

- 10) What are the criteria and mechanisms for Call Admission Control?
- 11) What are the criteria and mechanisms for Call Authentication?
- 12) How should Service Providers exchange and clear call detail records?
- 13) What information must be collected for accounting purposes?
- 14) Is there a recommended behavior for interconnecting fee-based and free voice services?
- 15) What are the recommended practices and/or solutions for SIP services utilizing IPv6 and/or DNS-Sec?
- 16) What are the base call services defined for a SIP interconnect?