# The VoIP Peering Puzzle�Part 2: The IETF SPEERMINT Requirements

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In our last tutorial on carrier technologies, we looked at some of the concepts and challenges of VoIP peering—where multiple enterprise locations, or possibly multiple autonomous networks, connect directly to each other using an Internet Protocol-based link, thereby bypassing the PSTN. We saw that for these two (or more) networks to be peers (or equals) a number of technical challenges need to be addressed.

These include:

- Interoperability between the peering networks
- Compatibility with signaling or call setup procedures
- Security
- Telephone-number-to-IP-address-database lookups, and others.

In this tutorial—and following ones—we will consider an architecture being developed by the IETF (Internet Engineering Task Force) that describes a big picture view of peering, and suggests how some of these challenges should be addressed.

In March 2006, the IETF, which develops the technical systems and standards for the Internet, created a new study 'Area,' called Real-Time Applications and Infrastructure to centralize all of the VoIP technology development (see http://www.ietf.org/html.charters/wg-dir.html#Real-time%20Applications%20and%20Infrastructure%20Area ).

This study Area is then divided into a number of working groups, which tackle a subset of that Area's research. Several of these working groups are investigating technologies directly relevant to VoIP peering. They include: the ENUM working group, which deals with telephone number mapping (see http://www.ietf.org/html.charters/enum-charter.html); the Session Initiation Protocol (SIP) working group, (see http://www.ietf.org/html.charters/sip-charter.html), the Signaling Transport (Sigtran) working group, which studies call setup procedures between IP networks and the PTSN (http://www.ietf.org/html.charters/sigtran-charter.html), and the recently formed Session PEERing for Multimedia INTerconnect, or SPEERMINT working group (who said that a bunch of engineers don't have any clever brain cells!), which "focuses on architectures to identify, signal, and route delay-sensitive (real-time) communication sessions" (see http://www.ietf.org/html.charters/speermint-charter.html).

The SPEERMINT working group has produced several Internet Draft documents that are relevant to our investigation into VoIP Peering. (It should be noted that Internet Drafts, as the name implies, are subject to revision, until such time that they are approved by the Working Group and other IETF authorities as Request for Comments (RFC) documents.) Two of these documents provide some very useful background information into how the IETF views SIP peering. The Terminology document (http://www.ietf.org/internet-drafts/draft-ietf-speermint-terminology-06.txt) defines some of the terms that are used to describe the interaction between SIP-based peers.

Of special note is the IETF's definition of peering:

"negotiation of reciprocal interconnection arrangements, settlement-free or otherwise, between operationally independent service providers."

In addition, there is an interesting distinction made between two different types of peering:

- Layer 3 peering: the interconnection of two service providers' networks for the purposes of exchanging IP packets which are destined for one (or both) of the peer's networks. Layer 3 peering is generally agnostic to the IP payload, and is frequently achieved using a routing protocol such as the Border Gateway Protocol (BGP) to exchange the required routing information.
- Layer 5 (Session) peering: the interconnection of two service providers for the purposes of routing real-time (or quasi-real time) secure call signaling between their respective customers using SIP methods.

In other words, when the SIP protocol in added to the mix, the peering relationships are necessarily more complex, as call signaling (call establishment and teardown) becomes part of the protocol operations.

The Requirements document (http://www.ietf.org/internet-drafts/draft-ietf-speermint-requirements-01.txt) describes high-level guidelines and general requirements for session peering; which could be applicable to any type of multimedia session peering, including VoIP, video telephony, and instant messaging. These requirements include: call addressing data, including Domain Name Service (DNS) and

Electronic Numbering (ENUM); the core SIP specifications that must be present for successful SIP calls; media-related requirements, such as compatible codecs; plus security aspects that should be addressed.

So is peering looking a little more complicated than you might have initially thought? Stay tuned, as our next tutorial will continue our exploration of the technical aspects of VoIP peering, and investigate the details of the SPEERMINT architecture.



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Author's Biography
Mark A. Miller, P.E. is President of DigiNet ® Corporation, a Denver-based consulting engineering firm. He is the author of many books on networking technologies, including Voice over IP Technologies, and Internet Technologies Handbook, both published by John Wiley & Sons.