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SPEERMINT Peering Architecture
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Abstract

This document defines the SPEERMINT peering architecture, its functional components and peering interface functions. It also

describes the steps taken to establish a session between two peering domains in the context of the functions defined.

Conventions used in this document

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 RFC-2119[1]

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1. Introduction

The objective of this document is to define a reference peering architecture in the context of Session PEERing for Multimedia INTerconnect (SPEERMINT). In this process, we define the peering reference architecture (reference, for short), it's functional components, and peering interface functions from the perspective of a SIP Service provider's (SSP) network.

This architecture allows the interconnection of two SSPs in layer 5 peering as defined in the SPEERMINT Requirements [13] and Terminology [12] documents.

Layer 3 peering is outside the scope of this document. Hence, the figures in this document do not show routers so that the focus is on Layer 5 protocol aspects.

This document uses terminology defined in the SPEERMINT Terminology document [12], so the reader should be familiar with all the terms defined there.

2. Network Context

Figure 1 shows an example network context. Two SSPs can form a Layer 5 peering over either the public Internet or private Layer3 networks. In addition, two or more providers may form a SIP (Layer 5) federation [13] on either the public Internet or private Layer 3 networks. This document does not make any assumption whether the SIP providers directly peer to each other or through Layer 3 transit network as per use case of [16].

Note that Figure 1 allows for the following potential SPEERMINT peering scenarios:

- Enterprise to Enterprise across the public Internet
- Enterprise to SSP across the public Internet

- SSP to SSP across the public Internet
- Enterprise to enterprise across a private Layer 3 network
- Enterprise to SSP across a private Layer 3 network
- SSP to SSP across a private Layer 3 network

The members of a federation may jointly use a set of functions such as location function, signaling function, media function, ENUM database or SIP Registrar, SIP proxies, and/or functions that synthesize various SIP and non-SIP based applications. Similarly, two SSPs may jointly use a set of functions. The functions can be either public or private.

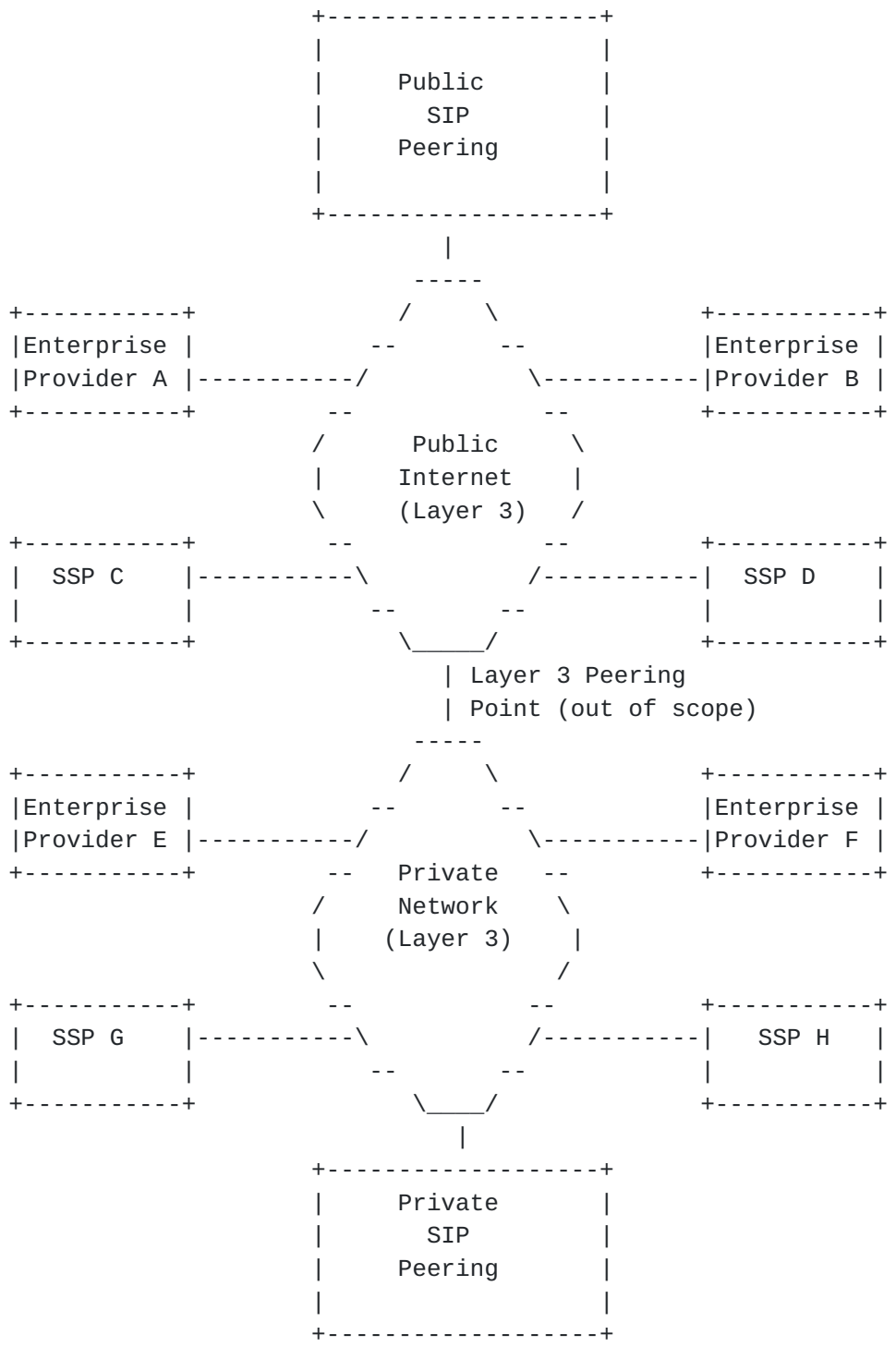


Figure 1: SPEERMINT Network Context

3. Procedures

This document assumes that in order for call to be establish from a UAC end user in the initiating peer's network to a UAS in the receiving peer's network the following steps are taken:

1. The analysis of the target address.

    . If the target address represents an intra-SSP resource, the behavior is out-of-scope with respect to this draft.

2. the determination of the target SSP,

3. the determination of the SF next-hop in the target SSP,

4. the enforcement of authentication and potentially other policies,

5. the determination of the UAS,

6. the session establishment,

7. the transfer of media which could include voice, video, text and others,

8. and the session termination.

The originating SSP would likely perform steps 1-4, and the terminating SSP would likely perform steps 4-5.

In the case the target SSP is different from the terminating SSP it would repeat steps 1-4. This is reflected in Figure 2 that shows the target SSP with its own peering functions.

4. Reference SPEERMINT Architecture

Figure 2 depicts the SPEERMINT architecture and logical functions that form the peering between two SSPs.

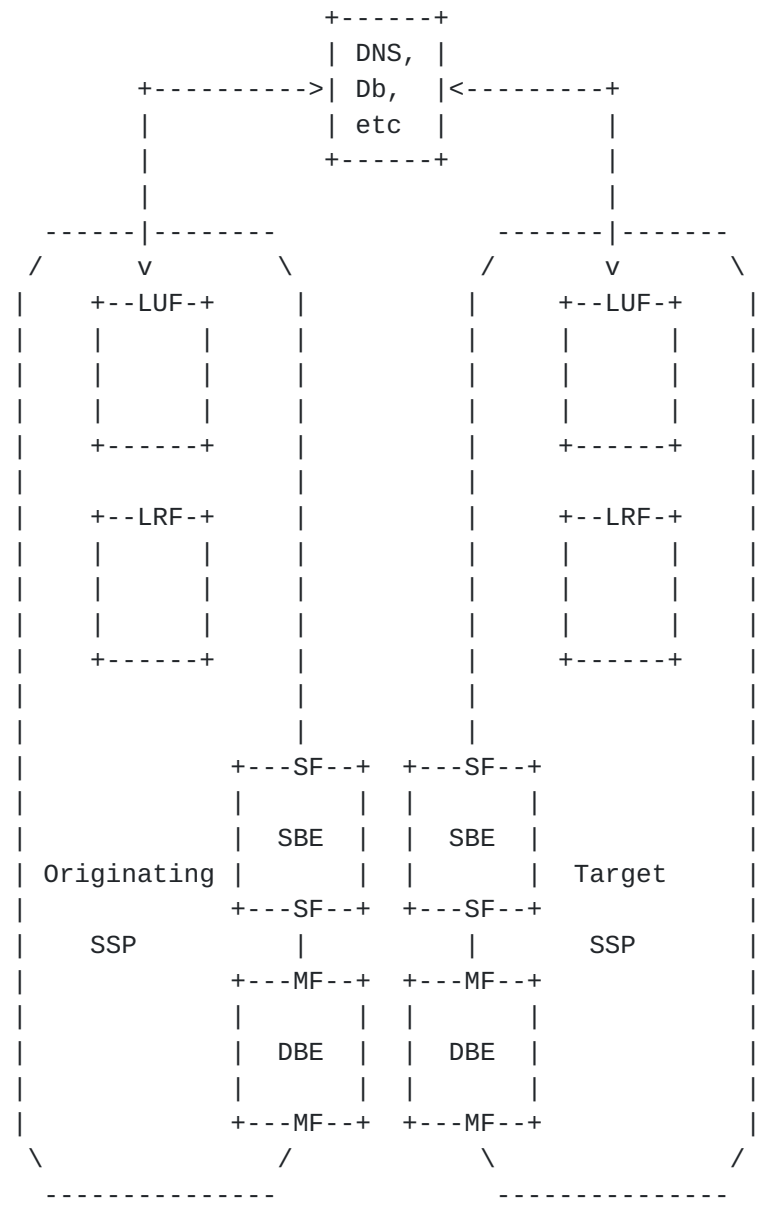


Figure 2: Reference SPEERMINT Architecture

The procedures presented in section 3 are implemented by a set of peering functions:

The Look-Up Function (LUF) provides a mechanism for determining for a given request the target domain to which the request should be routed.

The Location Routing Function (LRF) determines for the target domain of a given request the location of the SF in that domain and

optionally develops other Session Establishment Data (SED) required to route the request to that domain.

Signaling Function (SF): Purpose is to perform SIP call routing, to optionally perform termination and re-initiation of call, to optionally implement security and policies on SIP messages, and to assist in discovery/exchange of parameters to be used by the Media Function (MF).

Media Function (MF): Purpose is to perform media related function such as media transcoding and media security implementation between two SIP providers.

The intention of defining these functions is to provide a framework for design segmentation and allow each one to evolve independently.

5. Recommended SSP Procedures

This section describes the functions in more detail and provides some recommendations on the role they would play in a SIP call in a Layer 5 peering scenario.

Some of the information in the chapter is taken from [14] and is put here for continuity purposes.

5.1. Originating SSP Procedures

5.1.1. The Look-Up Function (LUF)

Purpose is to determine the SF of the target domain of a given request and optionally develop Session Establishment Data (SED) [12].

5.1.1.1. Target address analysis

When the initiating SSP receives a request to communicate, it analyzes the target URI to determine whether the call needs to be terminated internally or externally to its network. The analysis method is internal to the SSP; thus, outside the scope of SPEERMINT. Note that the SSP is free to consult any manner of private data sources to make this determination.

If the target address does not represent a resource inside the initiating SSP's administrative domain or federation of domains, the initiating SSP resolves the call routing data by using the Location Routing Function (LRF).

For example, if the request to communicate is for an im: or pres: URI type, the initiating peer follows the procedures in [8]. If the highest priority supported URI scheme is sip: or sips:, the initiating peer skips to SIP DNS resolution in Section 5.1.3. Likewise, if the target address is already a sip: or sips: URI in an external domain, the initiating peer skips to SIP DNS resolution in Section 5.1.2.2.

If the target address corresponds to a specific E.164 address, the peer may need to perform some form of number plan mapping according to local policy. For example, in the United States, a dial string beginning "011 44" could be converted to "+44", or in the United Kingdom "00 1" could be converted to "+1". Once the peer has an E.164 address, it can use ENUM.

5.1.1.2. User ENUM Lookup

If an external E.164 address is the target, the initiating peer consults the public "User ENUM" rooted at e164.arpa, according to the procedures described in RFC 3761. The peer must query for the "E2U+sip" enumservice as described in RFC 3764 [11], but MAY check for other enumservices. The initiating peer MAY consult a cache or alternate representation of the ENUM data rather than actual DNS queries. Also, the peer may skip actual DNS queries if the initiating peer is sure that the target address country code is not represented in e164.arpa. If a sip: or sips: URI is chosen the peer skips to Section 5.1.6.

If an im: or pres: URI is retrieved based on an "E2U+im" [10] or "E2U+pres" [9] enumserver, the peer follows the procedures for resolving these URIs to URIs for specific protocols such a SIP or XMPP as described in the previous section.

5.1.1.3. Infrastructure ENUM lookup

Next the initiating peer checks for a carrier-of-record in a carrier ENUM domain according to the procedures described in [12]. As in the previous step, the peer may consult a cache or alternate representation of the ENUM data in lieu of actual DNS queries. The peer first checks for records for the "E2U+sip" enumservice, then for the "E2U+pstn" enumservice as defined in [21]. If a terminal record is found with a sip: or sips: URI, the peer skips to Section 5.1.2.2. , otherwise the peer continues processing according to the next section.

5.1.2. Location Routing Function (LRF)

The LRF of an Initiating SSP analyzes target address and discovers the next hop signaling function (SF) in a peering relationship. The resource to determine the SF of the target domain might be provided by a third-party as in the assisted-peering case.

5.1.2.1. Routing Table

If there is no user ENUM records and the initiating peer cannot discover the carrier-of-record or if the initiating peer cannot reach the carrier-of-record via SIP peering, the initiating peer still needs to deliver the call to the PSTN or reject it. Note that the initiating peer may still forward the call to another SSP for PSTN gateway termination by prior arrangement using the routing table.

If so, the initiating peer may rewrite the Request-URI to address the gateway resource in the target SSP's domain and may forward the request on to that SSP using the procedures described in the remainder of these steps.

Alternatively to Request-URI re-writing, the initiating peer may populate the Route header with the address of the gateway resource in the target SSP's domain and forward the request on to that SSP using the procedures described in the remainder of these steps, but applied to the Route header.

5.1.2.2. SIP DNS Resolution

Once a sip: or sips: in an external domain is selected as the target, the initiating peer may apply local policy to decide whether forwarding requests to the target domain is acceptable. If so, the initiating peer uses the procedures in RFC 3263 [4] Section 4 to determine how to contact the receiving peer. To summarize the RFC 3263 procedure: unless these are explicitly encoded in the target URI, a transport is chosen using NAPTR records, a port is chosen using SRV records, and an address is chosen using A or AAAA records. Note that these are queries of records in the global DNS.

When communicating with another SSP, entities compliant to this document should select a TLS-protected transport for communication from the initiating peer to the receiving peer if available. Note that this is a single-hop requirement.

5.1.2.3. SIP Redirect Server

A SIP Redirect Server may help in resolving the current address of the next-hop SF in the target domain.

5.1.3. The Signaling Function (SF)

The purpose of signaling function is to perform routing of SIP messages, to optionally perform termination and re-initiation of a call, to optionally implement security and policies on SIP messages, and to assist in discovery/exchange of parameters to be used by the Media Function (MF).

The signaling function performs the routing of SIP messages. The optional termination and re-initiation of calls are performed by the signaling path border element (SBE).

Optionally, a SF may perform additional functions such as Session Admission Control, SIP Denial of Service protection, SIP Topology Hiding, SIP header normalization, and SIP security, privacy and encryption.

The SF can also process SDP payloads for media information such as media type, bandwidth, and type of codec; then, communicate this information to the media function. Signaling function may optionally communicate with the network to pass Layer 3 related policies [10]

5.1.3.1. Establishing a Trusted Relationship

Depending on the security needs and trust relationship between SSPs, different security mechanism can be used to establish SIP calls. These are discussed in the following subsections.

5.1.3.1.1. TLS connection

Once a transport, port, and address are found, the initiating SSP will open or find a reusable TLS connection to the peer. The procedures to authenticate the SSP's target domain is specified in [24]

5.1.3.1.2. IPSec

In certain deployments, the use of IPSec between the signaling functions of the originating and terminating domains can be used as a security mechanism instead of TLS.

5.1.3.1.3. Co-Location

In this scenario, the SFs are co-located in a physically secure location and/or are members of a segregated network. In this case messages between the originating and terminating SSPs would be sent as clear text.

5.1.3.2. Sending the SIP request

Once a trust relationship between the peers is established, the initiating peer sends the request.

5.1.3.2.1. TLS

If the trust relationship was established through TLS, the initiating peer can optionally verify and assert the sender's identity using the SIP Identity mechanism.

In addition, new requests should contain a valid Identity and Identity-Info header as described in [12]. The Identity-Info header must present a domain name that is represented in the certificate provided when establishing the TLS connection over which the request is sent. The initiating peer should include an Identity header on in-dialog requests as well if the From header field value matches an identity the initiating peer is willing to assert.

5.2. Terminating SSP Procedures

5.2.1. The Location Function (LF)

5.2.1.1. Publish ENUM records

The receiving peer should publish "E2U+SIP" and "E2U+pstn" records with sip: or sips: URIs wherever a public carrier ENUM root is available. In the event that a public root is not available, a publishing to a common ENUM registry with the originating peer will suffice.

This assumes that the receiving peer wants to peer by default. When the receiving peer does not want to accept traffic from specific initiating peers, it may still reject requests on a call-by-call basis.

5.2.1.2. Publish SIP DNS records

To receive peer requests, the receiving peer must ensure that it publishes appropriate NAPTR, SRV, and address (A and/or AAAA) records in the LF relevant to the originating peer's SF.

5.2.1.3. Subscribe Notify

A policy notification function may also be optionally implemented by dynamic subscribe, notify, and exchange of policy information and feature information among SSPs [21].

5.2.2. Signaling Function (SF)

5.2.2.1. TLS

When the receiving peer receives a TLS client hello, it responds with its certificate. The target SSP certificate should be valid and rooted in a well-known certificate authority. The procedures to authenticate the SSP's originating domain are specified in [24].

The terminating SF verifies that the Identity header is valid, corresponds to the message, corresponds to the Identity-Info header, and that the domain in the From header corresponds to one of the domains in the TLS client certificate.

5.2.2.2. Receive SIP requests

Once a trust relationship is established, the receiving peer is prepared to receive incoming SIP requests. For new requests (dialog forming or not) the receiving peer verifies if the target (request-URI) is a domain that for which it is responsible. For these requests, there should be no remaining Route header field values. For in-dialog requests, the receiving peer can verify that it corresponds to the top-most Route header field value.

The receiving peer may reject incoming requests due to local policy. When a request is rejected because the initiating peer is not authorized to peer, the receiving peer should respond with a 403 response with the reason phrase "Unsupported Peer".

5.3. Target SSP Procedures

5.3.1. Signaling Function (SF)

5.3.1.1. TLS

When the receiving peer receives a TLS client hello, it responds with its certificate. The target SSP certificate should be valid and rooted in a well-known certificate authority. The procedures to authenticate the SSP's originating domain are specified in [24].

If the requests should contain a valid Identity and Identity-Info header as described in [12] the target SF verifies that the Identity header is valid, corresponds to the message, corresponds to the Identity-Info header, and that the domain in the From header corresponds to one of the domains in the TLS client certificate.

5.3.1.2. Receive SIP requests

The procedures of the SF of the target SSP are the same as the ones described in section 5.2.2.2 with the addition that it might establish a connection to another target SSP, and in this case use the procedures recommended to an originating SSP (section 5.1).

5.4. Media Function (MF)

The purpose of the MF is to perform media related functions such as media transcoding and media security implementation between two SSPs.

An Example of this is to transform a voice payload from one codec (e.g., G.711) to another (e.g., EvRC). Additionally, the MF may perform media relaying, media security, privacy, and encryption.

5.5. Policy Considerations

In the context of the SPEERMINT working group when two SSPs peer, there MAY be a desire to exchange peering policy information dynamically. There are specifications in progress in the SIPPING working group to define policy exchange between an UA and a domain [23] and providing profile data to SIP user agents [24] These considerations borrow from both.

Following the terminology introduced in [12], this package uses the terms Peering Session-Independent and Session-Specific policies in the following context.

- Peering Session-Independent policies include Diffserv Marking, Policing, Session Admission Control, and domain reachabilities, amongst others. The time period between Peering Session-Independent policy changes is much greater than the time it takes to establish a call.

- Peering Session-Specific polices includes supported connection/call rate, total number of connections/calls available, current utilization, amongst others. Peering Session-specific policies can change within the time it takes to establish a call.

Likewise, but orthogonal to session dependency, an SSP may have policies that may be peer-dependent or peer-independent. That is, the session-dependent and session-independent policies may by further sub-divided and modified by additional controls that depend on which peer SSP or federation with which communications is being established.

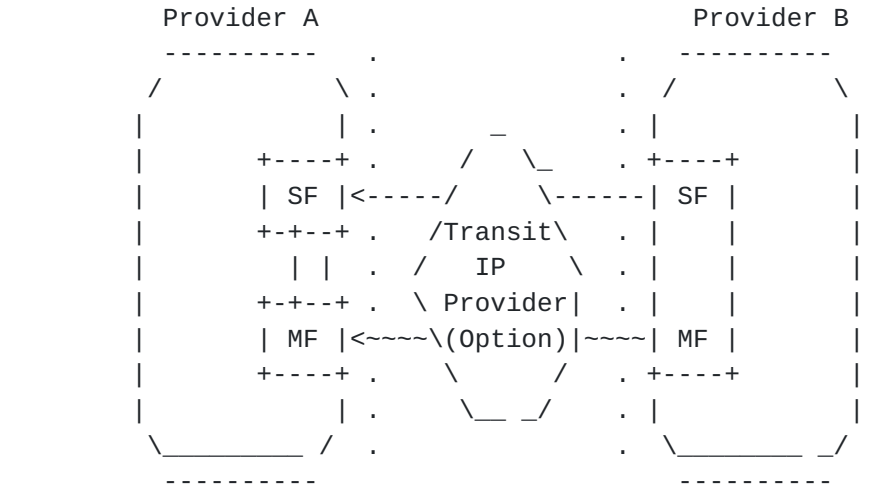
6. Call Control and Media Control Deployment Options

The peering functions can be deployed along the following two dimensions depending upon how the signaling and the media functions along with IP layer are implemented:

Composed or Decomposed: Addresses the question whether the media must flow through the same physical and geographic elements as SIP dialogs and sessions.

Centralized or Distributed: Addresses the question whether the logical and physical interconnections are in one geographical location or distributed to multiple physical locations on the SSP's network.

In a composed model, SF and MF functions are implemented in one peering logical element.



--- Signal (SIP)
~~~ Bearer (RTP/IP)
... Scope of peering

Figure 3: Decomposed v. Collapsed Peering

The advantage of a collapsed peering architecture is that one-element solves all peering issues. Disadvantage examples of this architecture are single point of failure, bottleneck, and complex scalability.

In a decomposed model, SF and MF are implemented in separate peering logical elements. SFs are implemented in a proxy and MFs are implemented in another logical element. The scaling of signaling versus scaling of media may differ between applications. Decomposing allows each to follow a separate migration path.

This model allows the implementation of M:N model where one SF is associated with multiple peering MF and one peering MF is associated with multiple SFs. Generally, a vertical protocol associates the relationship between a SF and a MF. This architecture reduces the potential of a single point of failure. It allows separation of the policy decision point and the policy enforcement point. An example of disadvantages is the scaling complexity because of the M:N relationship and latency due to the vertical control messages between entities.
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7. Address space considerations

Peering must occur in a common IP address space, which is defined by the federation, which may be entirely on the public Internet, or some private address space. The origination or termination networks may or may not entirely be in the same address space. If they are not, then a network address translation (NAT) or similar function may be needed before the signaling or media is presented correctly to the federation. The only requirement is that all associated entities across the peering interface are reachable.

8. Security Considerations

In all cases, cryptographic-based security should be maintained as an optional requirement between peering providers conditioned on the presence or absence of underlying physical security of peer connections, e.g. within the same secure physical building.

In order to maintain a consistent approach, unique and specialized security requirements common for the majority of peering relationships, should be standardized within the IETF. These standardized methods may enable capabilities such as dynamic peering relationships across publicly maintained interconnections.

TODO: Address RFC-3552 BCP items.

9. IANA Considerations

There are no IANA considerations at this time.

10. Acknowledgments

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A portion of this draft is taken from [14] with permission from the author R. Mahy. The other important contributor is Otmar Lendl.

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