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Use of DNS SRV and NAPTR Records for SPEERMINT
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Status of this Memo

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Abstract

The objective of this document is to specify the Best Current Practice (BCP) adopted by a service provider providing multimedia communication services such as Voice over Internet Protocol(VoIP) in order to locate another service provider to peer with in the context of Session PEERing for Multimedia INTerconnect.

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1. Introduction

A service provider needs to identify the ingress Session Initiation Protocol (SIP) (RFC 3261 [1]) server of a peering network before it can signal and route SIP based real-time communication sessions. This function of locating the ingress SIP server of peering network is typically performed by the egress SIP server of the service provider originating the SIP session. Also, the ingress server in the peering network needs to locate the originating service provider egress server in situations where the peering connection to it gets terminated after receiving the SIP requests or if the egress SIP server of originating service provider fails. The SIP servers at originating and peering side use the DNS procedures, using both SRV [2] and NAPTR [3] records, in order to locate each other.

2. Terminology

In this document, the key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" are to be interpreted as described in RFC 2119 [2] and

indicate requirement levels for compliant implementations.

3. Session Peering Setup

SIP systems are represented by user agents (UA). The diagram below shows the case of direct peering where a user agent (UA1), hosted by

a service provider SP1, initiates a SIP session to a User Agent (UA2), hosted by service provider SP2. The egress SIP server of SP1 is a SIP signaling path border element (SBE) as defined in section 3 of[6], called SBE1, that interfaces with session peering service provider SP2. The SIP session initiated by UA1 is received by this network element SBE1. The resource to which the SIP request needs to be routed by SBE1 is identified by a SIP or SIPS URI. This could be the SIP URI of UA2 found in the Request-URI of the SIP request received by SBE1, or the next hop from SBE1 found in the topmost Route header of SIP request. In order to determine the resource to route the request to, SP1 MAY make use of ENUM [4] lookup services or an internal lookup to determine the SIP URI of the resource. This lookup MAY be performed by SBE1 or another network element of SP1.

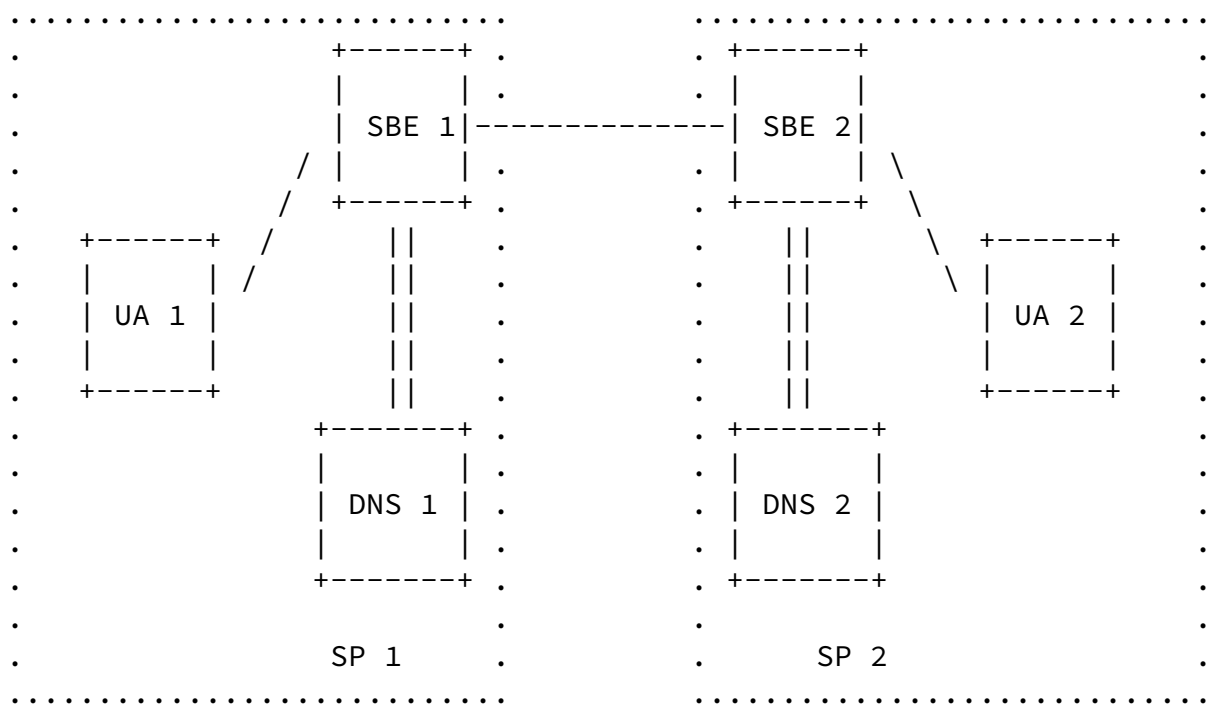


Figure 1: Logical Peering Scenario (direct peering)

In order to route the SIP request to this resource in SP2, SBE1 needs to determine the ingress SIP signaling path border element for SP2, called SBE2, by resolving the SIP or SIPS URI in DNS. SBE1 makes use of the NAPTR and DNS SRV mechanism defined in [5] to determine the IP address, port, and transport protocol for peering with the SP2 ingress SIP proxy server (i.e. SBE2). SBE1 and SBE2 which are involved in the session peering, support a set of protocols and have list of preferences for these protocols. UDP, TCP and TLS MUST be supported by these proxies.

As a best current practice, SBE1 and SBE2 SHOULD be deployed in a highly scalable and highly available manner, such as a cluster of

servers. These servers are of different prioritization and weight, to ensure capacity-based load balancing.

The figure below shows the case of indirect/transit peering where SBE2 is the ingress SIP server of a transit service provider. The mechanism to locate SBE2 is the same as described for direct peering scenario.

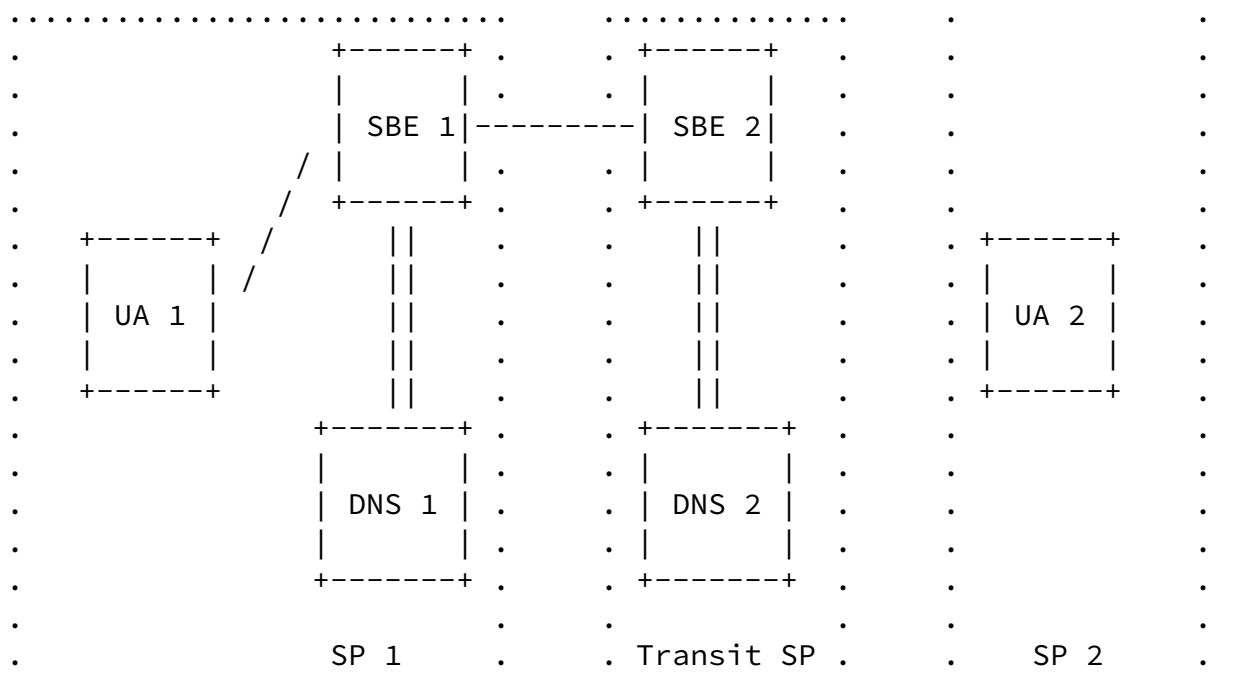
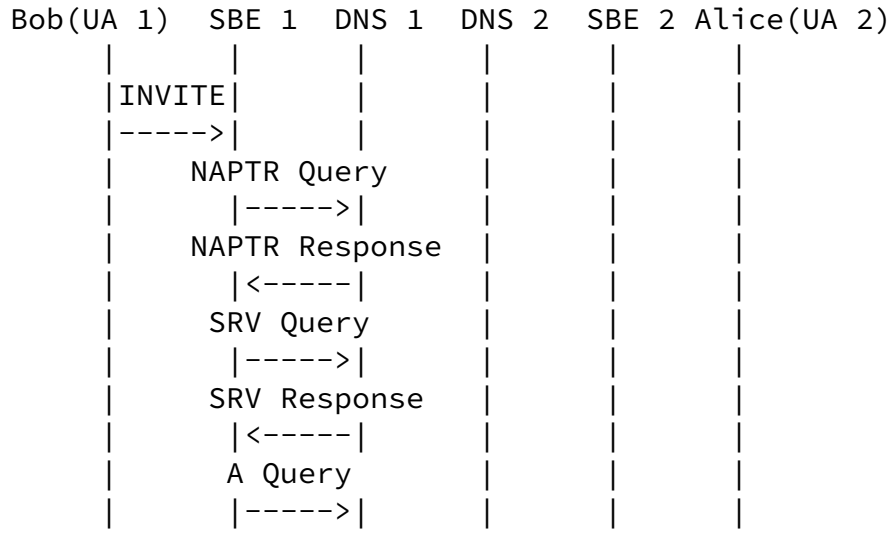


Figure 1: Logical Peering Scenario (indirect peering)

The figure below shows a high level SIP call flow setting up a direct SIP peering session between SP1 and SP2. In this call flow a VoIP session is eastablished between a caller, Bob (sip:bob@sp.com), in SP1 and callee, Alice(sip:alice@sp2.com), in SP2 using SIP INVITE request. All SIP signaling MUST go through the SBE1 and SBE2 as these are the ingress and egress points in SP1 and SP2 network.



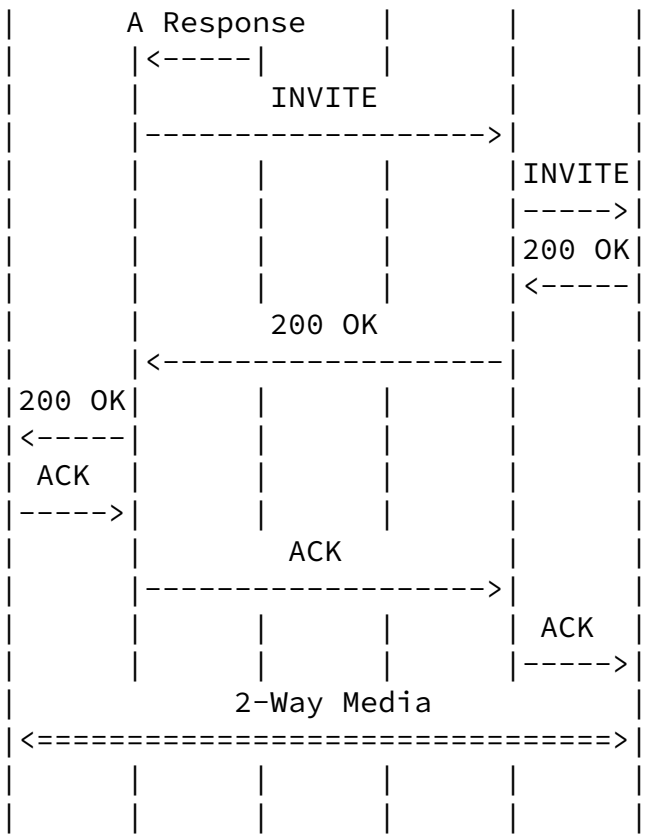


Figure 2: Example Call Flow

The target, to which the request is sent, is determined by SBE1 as follows:

3.1 TARGET Determination

The target resource is identified with a SIP or SIPS URI. This is the URI in the Route header, if present, or the URI from the request URI of the SIP request received by SBE1.
The host value of the hostport component of the URI is the TARGET. This TARGET is the domain to be contacted. The NAPTR/SRV/A lookup as described in the following section should be skipped if transport/port/ip address is already specified for the target URI.

3.2 NAPTR Lookup

Next the SBE1 determines the transport protocol of the TARGET by performing a NAPTR query for the TARGET. NAPTR processing as described in [3] will result in the discovery of the most preferred transport protocol of a server instance of SBE2 and SRV records.

Considering our example call flow setup above, SBE1 wishes to resolve sip:alice@sp2.com and performs a NAPTR query for that TARGET domain sp2.com, and the following NAPTR records are returned:

```
;           order pref flags service     regexp  replacement
   IN NAPTR 50   50  "s"  "SIPS+D2T"     ""    _sips._tcp.sp2.com
   IN NAPTR 90   50  "s"  "SIP+D2T"      ""    _sip._tcp.sp2.com
   IN NAPTR 100  50  "s"  "SIP+D2U"      ""    _sip._udp.sp2.com
```

DNS MUST return at least three records - one with "SIP+D2T", one with "SIP+D2U" and one with "SIPS+D2T" service type for the case of direct and indirect peering (section 4.3 in [6]). For indirect (transit) peering (section 4.4 in [6]) since domain validation as specified in section 26.3.2.2 of [1] for TLS at layer 5 will not work, SIPS over TLS cannot be used.

3.3 SRV Lookup

Depending on what transport protocols SBE1 supports, SBE1 selects one from the preference list of NAPTR results and performs the SRV lookup to obtain a list of available server instances for SBE2. TLS SHOULD be the preferred transport protocol for peering between SBE1 and SBE2.

In our example SBE1 uses TCP, the SRV lookup for _sip._tcp.sp2.com would return list of available servers :

```
;;          Priority Weight Port   Target
    IN SRV  0        1      5060   server1.sp2.com
    IN SRV  0        2      5060   server2.sp2.com
```

Alternatively, if no NAPTR records are found, then SBE1 uses the preferred transport protocol and issues an SRV query for that specific transport using "sips" for SIPS URI and SIP URI with TLS and "sip" for SIP URI as the SRV domain prefix.

In our example, SBE1 prefers to use TCP and target SIP URI of SP2 is sip:alice@sp2.com, it sends a SRV query for _sip._tcp.sp2.com.

The SRV responses MAY also include A records with it.

3.4 Using SRV Results

If A records are not returned with the SRV responses, procedure from RFC 2782 describes how to use and interpret the results obtained from the SRV query. The target entry of the SRV RRs is looked up by querying the DNS for address records. If the SRV response from DNS includes A records with it, it will cut down on round trips and lookup of DNS again for target entry. On determining the transport protocol, service, port and address record from the SRV RRs as described above, the SBE1 will try to connect to the (protocol, address, service). Once the connection is established to an available instance of SBE2, SBE1 sends the SIP request to SBE2. SBE1 MUST act in a stateful manner and any retransmission of SIP requests for a specific SIP transaction, including ACKS for non-2xx response or CANCEL for that SIP transaction MUST go to the same server instance of SBE2.

When SBE1 sends the SIP request to SBE2, it SHOULD set the sent-by parameter of the topmost Via header in the SIP request to a domain that identifies SBE1. It MUST NOT specify the port.

4. High Availability

High Availability is ensured by detecting failures in the ability to connect to SBE1 and SBE2 server instances. In the event of a failure, when SBE1 tries to send SIP INVITE to SBE2, the following failures could occur:

4.1 SBE1 Fails to Reach SBE2

A 503 error response is reported by the transaction layer, or failure can occur at the transport layer due to TCP disconnect in connection,

ICMP error in UDP or time out at transport layer or SIP layer timeout when its not receiving any SIP response. In such situations, SBE1 tries a new SIP request transaction to the next available server instance of SBE2 as determined by SRV RRs entry. The SIP T1 timer on SBE1 SHOULD be configurable with a upper limit value of 500ms. A shorter value of T1, say 100ms, reflects a faser failover support.

4.2 SBE2 Fails to Reach SBE1

Failure may also occur after the request is received by SBE2 from SBE1 due to closure of the transport connection the request came in on at SBE2, before the response can be sent back to SBE1. In this situation, SBE2 uses the domain value present in the 'sent-by' parameter in the top most Via header of the received SIP INVITE, and queries for SRV records at this domain name using the service identifier "_sips" if the Via transport is "TLS", "_sip" otherwise. The sorted list of SRV RRs are obtained and used as described in [2] to send the response back to SBE1. If the topmost element in the list of server instances of SBE1 fails, the next available one is tried.

[FOR NEXT REV: SHOULD WE ADD CALL FLOW FOR FAILURE SCENARIO DESCRIBED IN 4.1 AND 4.2]

5. Caching/TTL

5.1 Caching

SBE SHOULD use caching of DNS results to eliminate unnecessary DNS queries.

5.2 TTL

SRV RRs have a TTL value based on which the SBE1 caches the entry for that duration, if it supports caching, and any further requests to the same TARGET domain are delivered to the cached server instance. The TTL recommended for SRV is about 1 hr. The TTL for NAPTR is much higher, about 1 day (24hrs) since the NAPTR records do not vary that often as compared to SRV.

6. Acknowledgements

Special thanks go to Yiu Lee for his valuable input to this document.

7. Security Considerations

This document introduces no new security considerations.

8. IANA Considerations

This document creates no new requirements on IANA namespaces [RFC2434].

9. References

9.1 Normative References

[1] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M. and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002.

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[6] Meyer, D., "SPEERMINT Terminology", draft-ietf-speermint-terminology-12, August 2007.

9.2 Informative References

[RFC2434] Narten, T. and H. Alvestrand, "Guidelines for Writing an IANA Considerations Section in RFCs", BCP 26, RFC 2434, October 1998.

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