

SIP Trunking

SIP Connectivity for Inbound and Outbound Calls

CONFIGURING INBOUND CONNECTIVITY

- We strongly recommend using DNS SRV (service) records to deliver calls to your UC infrastructure from Avaya SIP Trunking.

SRV records contain a list of destinations with priorities and weights for call delivery.

This allows for automated failover (endpoints with higher priority numbers are used only if lower priority number endpoints are unresponsive), and load balancing (Calls are distributed in proportion to weight). Use of DNS SRV records for call delivery also allows the system to change routing priorities and weights (i.e. for problems, emergencies, or maintenance).

The example on page xxx shows how to set-up multiple SRV records for a single domain call handling scenario for SIP.

Note: The customer is responsible for configuring the DNS SRV record(s). The full DNS SRV record names are [_sip._tcp.pbx.companyabc.com](#) (TCP signaling) or [_sip._tls.pbx.companyabc.com](#) (TLS). The customer must whitelist a set of IP addresses coming from the Avaya SIP Trunk for signaling.

- Assigning a Static IP Address

The disadvantage of using IP addresses for call delivery is that there is no failover or load balancing. This is a static route to a single server. However, on our portal customers have the flexibility to change the inbound route to a new IP address that takes effect immediately.

SIP SRV EXAMPLE AND BREAKDOWN

Service	Protocol	Name	TTL	Class	SRV	Priority	Weight	Port	Target
_sip	_tcp	pbx.companyabc.com	86400	IN	SRV	10	10	5060	pbx1.companyabc.com
_sip	_tcp	pbx.companyabc.com	86400	IN	SRV	10	10	5060	pbx2.companyabc.com
_sip	_tcp	pbx.companyabc.com	86400	IN	SRV	20	10	5060	pbx-bkup1.companyabc.com
_sip	_tcp	pbx.companyabc.com	86400	IN	SRV	20	10	5060	pbx-bkup2.companyabc.com

In this example, during normal operation, whenever a call needs to be directed to pbx.companyabc.com, it will be sent to either pbx1.companyabc.com or pbx2.companyabc.com. If neither of those are accessible, it will route calls to the last 2. Note that if the Target is a name such as shown above, you will also be responsible for creating the A records to match.

SAMPLE

This example shows how to change the Inbound Route on our Portal. Open your Numbers, select the

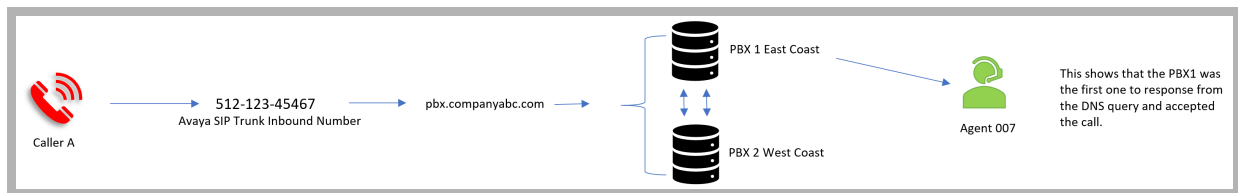
number and change the WebLink to the new IP or DNS name (starting with sip://).

The screenshot shows the configuration page for a number. The 'WebLink' field is highlighted with a red box and contains the value 'sip://pbx.companyabc.com:5060;Transport-TCP'. Other fields include 'Request URL' (sip://), 'Webhooks', and 'SMS Configuration'.

Company ABC has 2 PBX servers across North America and is using DNS SRV.

This example enables an active-active call delivery scenario so that Avaya SIP Trunking will deliver calls round-robin and which PBX will answer first. The DNS Admin can adjust the various requirements as required.

- **Pbx 1** – IP address 192.168.1.1 (Priority = 10 and Weight of 10)
- **Pbx 2** – IP address 192.168.1.2 (Priority = 10 and Weight of 10)



OUTBOUND CONNECTIVITY REQUIREMENTS

- Both Credential and IP Address Authentication are required. The site admin must provide the IP Address(es) where the call originates. These addresses must be whitelisted in order to allow outbound calling.
- The Dialplan should use the E164 format, or at least include the country code + full number.
- The Signaling IP address from our SBC will be provided if the customer site is not capable of supporting DNS SRV.
- Our SBCs will reply **200 OK** in response to **SIP OPTIONS**.

Note: The audio path for the SIP Trunking service goes directly from peering carriers to your endpoint (PBX Servers).