Configuration Guide Ingate SIParator/Firewall High Available Deployments with Amazon Web Services (AWS)

A how-to Guide

For the Ingate SIParator®/Firewalls using software release 6.2.1 or later

1 INTRODUCTION 3
2 GENERAL OVERVIEW 4
3 PROPOSED SOLUTION OVERVIEW 5
4 WHAT IS DNS SRV AND HOW DOES IT WORK? 7
4.1 DNS TECHNOLOGY 7
4.2 DNS ILLUSTRATED. 7
4.3 DNS SRV RECORDS STRUCTURE 8
4.3.1 Load Balancing 9
4.3.2 Failover 9
5 CONFIGURING DNS SRV ON AWS 10
5.1 CONFIGURING DNS SRV FOR PUBLIC HOSTED ZONES 10
5.2 DNS SRV AS AN ADDITIONAL OPTION FOR FAILOVER IN THE PRIVATE SUBNET 15
5.3 SIPARATOR CONFIGURATION FOR FAILOVER USING DNS SRV AND OTHER FEATURES (SIP TRUNKING USE CASE) 17
5.3.1 IP-PBX Failover 18
5.3.2 PBX Failover using DNS SRV for SIP Trunks 19
5.3.3 PBX Failover using Dual Homing for PBX Domain in the Trunk Group 20
5.3.4 SIP Trunks Failover. 22
5.3.5 Dual/Multi homed Carrier. 23
5.3.6 DNS SRV SIP Trunk (Carrier Side) 24
5.3.7 DNS SRV SIP Trunk (SIParator Side). 24
5.3.8 Twilio Redundancy Options 27
5.4 SIPARATOR CONFIGURATION FOR FAILOVER USING DNS SRV AND OTHER FEATURES (REMOTE USERS/ENDPOINTS USE CASE) 32
5.4.1 Using DNS SRV to become the SIP Proxy and Registrar with a Failover with 2 SIParators 32
5.5 PBX CROSS CONNECTION WITH SIPARATORS 36
6 REPLICATING AND SYNCHRONIZING SIPARATOR CONFIGURATIONS ACROSS ALL INSTANCES 36
6.1 HOW TO SYNCHRONIZE CONFIGURATION 38
6.1.1 Obtaining CLI file from the main SIParator 38
6.1.2 Manually modify CLI File: 39
6.1.3 Loading CLI file and use the GUI to make all needed adjustments 45
7 FINAL TIPS AND RECOMMENDATIONS 48
8 ADDITIONAL HELP OR SUPPORT 51

Version: Ingate SIParator/Firewall version 6.X

1 Introduction

This document aims to provide step by step guidance to design and deploy Highly Available Ingate SIParator instances in AWS/VPC. It leverages existing AWS VPC functionalities and Services which facilitate the goal of keeping operational continuity of your infrastructure.

Operational continuity will not be reachable by only applying these concepts to the SIParator, but also to the remaining infrastructure (IP-PBX, Soft switch, Routing gear, etc..) as they will need to be considered in the design. These considerations fall outside the scope of this guide.

SIParator appliances provide a way to implement High Availability by enabling Ingate's own SIParator Fail Over feature, pairing two identical Appliances into a cluster (see the Failover chapter in the Reference Guide) using one logical IP address. This functionality is not used in this document, as it is not available yet for AWS.

When configuring on AWS, the best way to proceed is by using DNS SRV records and some additional failover capabilities that will be shown in this document.

The two most common use cases when configuring SIParator are:

1) Provide remote access to users/endpoints to Telephony resources thru an SBC
2) Provide PSTN connectivity thru Service Providers by mediating SIP Trunks between the IPPBX and the Carrier.

A resilient implementation will be the one that provides minimum down time to both scenarios, users remotely connected willing to use Telephony Services and PSTN access

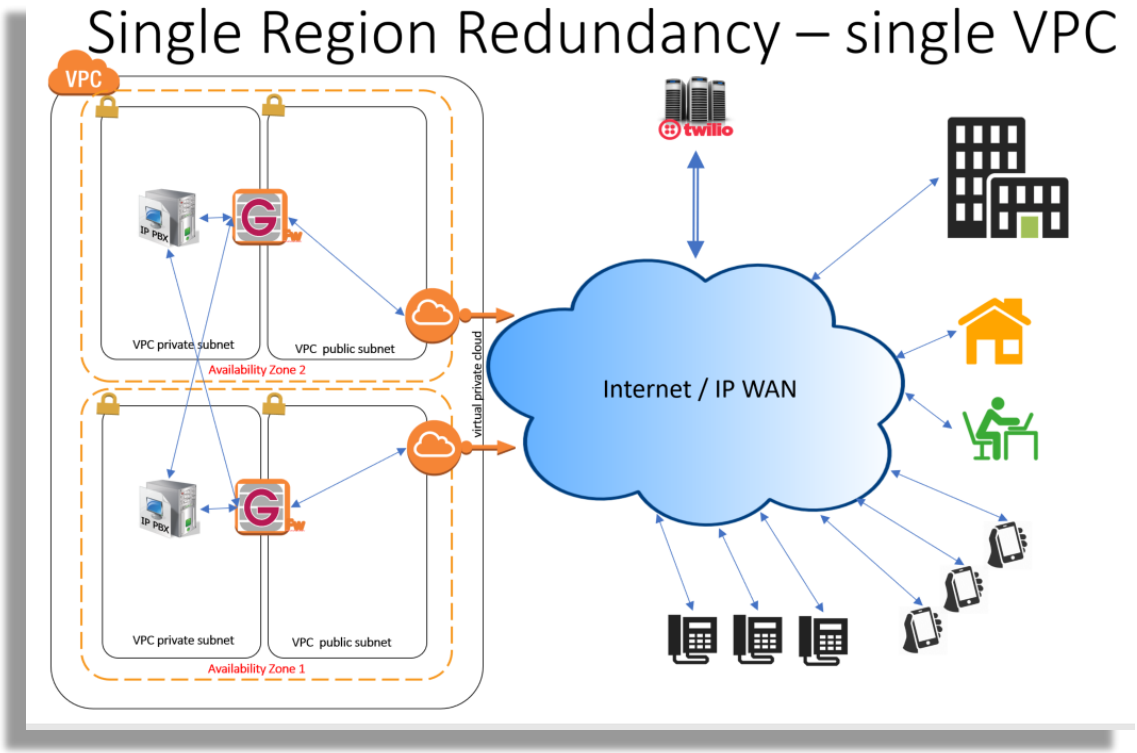
You will learn about the following topics:

1) How to use DNS Override (SIParator included feature) for remote extensions, and how it fits for redundant PBXes
2) How to use dual homing/domain support on SIParator Trunk Groups either facing the carrier, or facing the IP-PBX
3) How to use Route53 Service from AWS to build your DNS SRV records. (You can use your own managed DNS services)
4) How to use a Service Provider that already offers redundancy, accepts redundancy or supports DNS SRV

2 General Overview

In our case, we are going to implement a Redundant pair whereby each SIParator instance is in different AZ (Availability Zone) in the same VPC (Virtual Private Cloud). As each availability zone represents a physical data centre, we are protecting the solution not only against a potential Instance Failure, but also a full isolation of one of the data canters.

Note: this use case is totally different of "**SIParator Failover Pair"** feature explained in **"Ingate Reference Guide"** Chapter 13. Failover Pair feature uses heart beat between the 2 units using layer 2 protocols that are possible to implement in cloud environments such as AWS.



As shown in the figure, our two SIParators are located each one in a different AZ (separated Data Centers). Also, each one of them will have one interface in a Public Subnet (to face the Internet) and a second interface in a private subnet (to face the IP-PBX),

The interface facing the public subnet will have an Elastic IP (Public IP associated) for each SIParator.

One PBX will be located on the private subnet for each AZ (assuming they are configured in a failover fashion). And each SIParator will be able to reach any PBX. In this case, it is important to understand that All VPC subnets are always reachable from any other subnet as all of them belong to the same VPC.

As illustrated, we have the most common use cases represented here. We have PSTN connectivity using in our case TWILIO as the service provider and we have remote users in all possible fashions, including Remote Phones, Offices, Home Office, Softphones, etc.

Our goal in this case is to provide users with service continuity in the event one of the SIParators or PBX goes down.

DNS SRV, configured using Route53 (as explained in a later section)

3 Proposed Solution Overview

Here are the specific actions we will take to deploy a solution:

1) For remote users, we will use an FQDN (Fully Qualified Domain Name) on the public side of the SIParators, and such FQDN will be resolved using SRV Records in a Domain Server (Route53) to include Corresponding Elastics IPs of each SIParator. This will create a failover strategy to reach any of the SIParators

2) As our Domain Name Service to resolve external FQDN, we will use Route53 from AWS

3) We will use DNS Override feature in SIParator to route SIP Requests internally to one of the IPPBX with a similar approach to how DNS SRV is used

4) For SIP Trunks (PSTN), we will leverage Twilio's failover capabilities. For inbound (from PSTN to SBC). They provide two different options;

    a. Full support of DNS SRV records, so they will send traffic to an FQDN and SRV records will define the failover strategy,

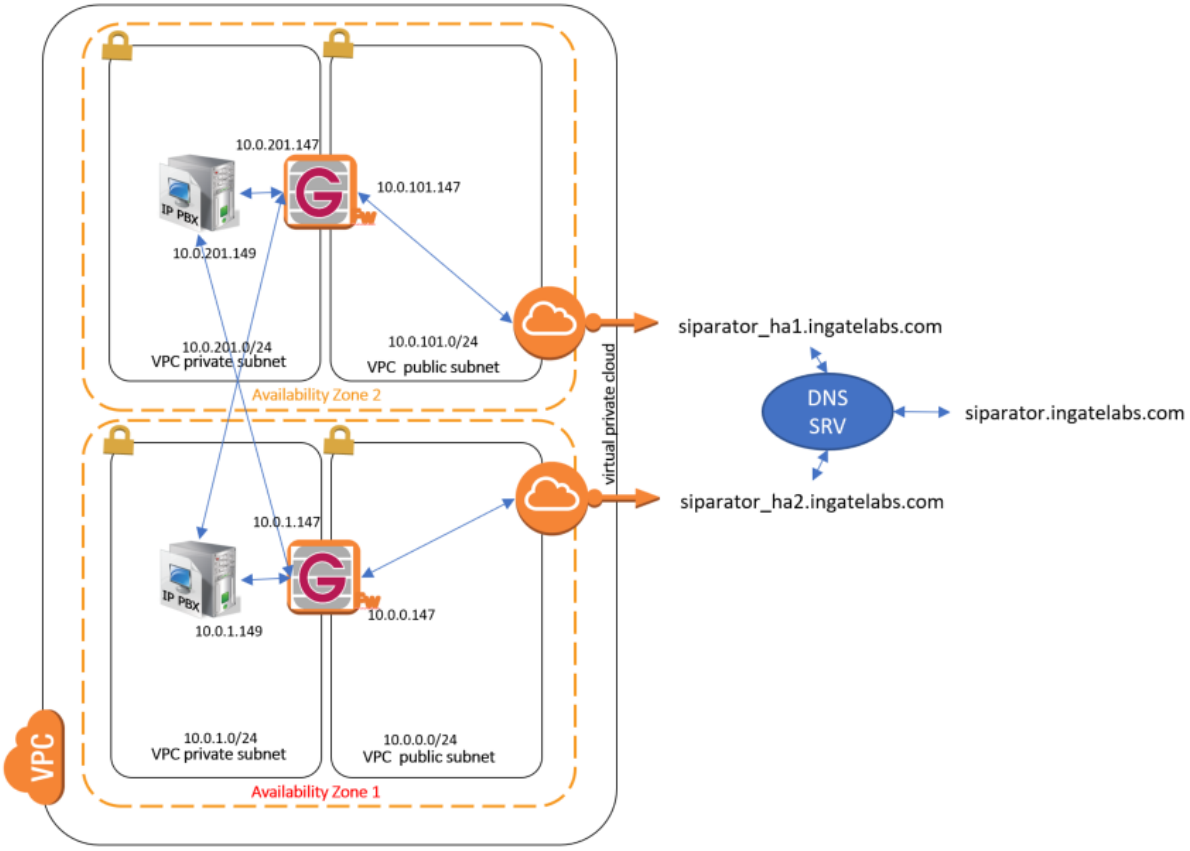
    b. To define more than one IP Address, to send traffic to, and failover from one to the other. This is useful when DNS SRV is not easy to implement.

5) Inbound traffic (Originated from PSTN) to be sent to IPPBX will be implemented by using redundant PBX domains supported in the Trunk Group Page

6) Outbound PSTN traffic from any of the two IPPBX to SBC can be implemented by accepting calls from any of the two IPPBX and routing to the appropriate SIP Trunk Group (Which already has a failover strategy as defined in 7)

7) For Outbound (From SBC to PSTN), Twilio Provides a unique FQDN for each trunk and it is resolved using DNS SRV. Here, Twilio already provides a failover strategy by default. It is important to know that in case the carrier does not have DNS SVR FQDN, but they can provide more than one IP for

failover, we can use the Dual Domain Feature for SIP Trunk configuration in the Ingate SIP Trunk Group.



What we are enabling here are the following failover strategies:

For Remote Users (Access use case)

1) Any remote user/extension uses a single registrar/proxy domain (siparator.ingatelabs.com), and via SRV records in case one SBC becomes unreachable, the other will take over operations. (siparator_ha1 and separator_ha2)

2) No matter which SBC is active, if any of the PBXs fails, the second PBX will be used instead.

For PSTN (trunking use case)

1) Traffic to the ITSP will be accepted from any of the SBCs. So, in case one fails, the second SBC will be capable of routing outbound calls.

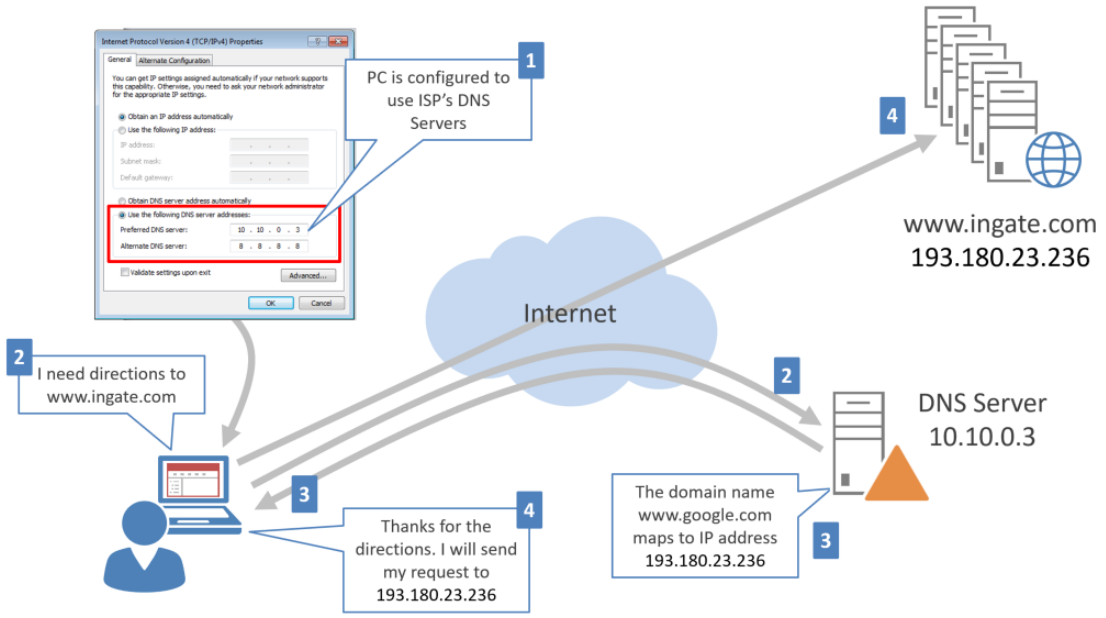
2) Both SBCs will be able to process inbound traffic. If one SBC becomes unreachable to the ITSP, the second SBC will be tried. (this failover is provided by the ITSP)

4 What is DNS SRV and how does it work?

4.1 DNS Technology

- DNS means Domain Name Service
- Is Used as a phonebook for the internet
- Internet works on IP addresses
- DNS Infrastructure allows you to remember something like www.ingate.com and instead is interpreted as the IP address 193.180.23.236
- DNS Infrastructure is a hierarchy of databases distributed across the internet (https://en.wikipedia.org/wiki/Domain_Name_System)

4.2 DNS Illustrated.



1) A Client device is configured to use a specific DNS Server (10.10.0.3)
2) The Client device seeks to access www.ingate.com, and asks the DNS Server
3) The DNS Server responds to the client with the IP address where www.ingate.com is located

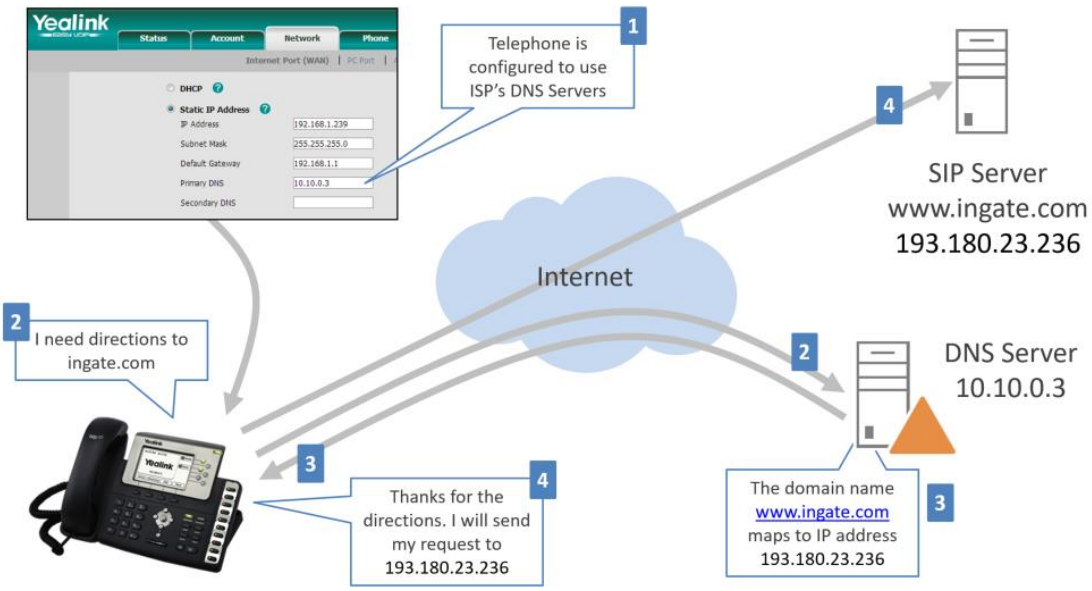
4) Now the Client sends the same request to the discovered IP address indicated by the DNS Server.

In SIP it is very similar. If I'm making a call the destination address might look something like this:

15554018146@ingate.com

For the call to proceed, the device needs to find where ingate.com is located.
It could be for instance a PBX, or a Soft switch, or even a proxy-server or SBC.

In a SIP call looks like:



- As mentioned, DNS is a massively distributed database, which is comprised of many records
- Database look ups and IP resolution take just a few milliseconds
- It is very reliable, as it is the foundation of the Internet
- DNS records are divided into types:
    - A Type: maps a single domain name to an IP address (1:1)
    - SRV Type: Service records. Useful for locating specific services (i.e. SIP) and multiple servers
    - Many others (MX, AAAA, etc.…)

4.3 DNS SRV records Structure

An SRV record follows this structure:

_Service._Proto.Name TTL Class SVR Priority Weight Port Target

Where:

- Service: the symbolic name of the desired service.
- Proto: the protocol of the desired service; this is usually either TCP or UDP.
- Name: the domain name for which this record is valid.
- TTL: standard DNS time to live field
- Class: standard DNS class field (this is always IN)
- Priority: the priority of the target host, lower value means more preferred
- Weight: A relative weight for records with the same priority
- Port: the TCP or UDP port on which the service is to be found
- Target: the canonical hostname of the machine providing the service

As an example, a query to **siparator.ingatelabx.com** would yield

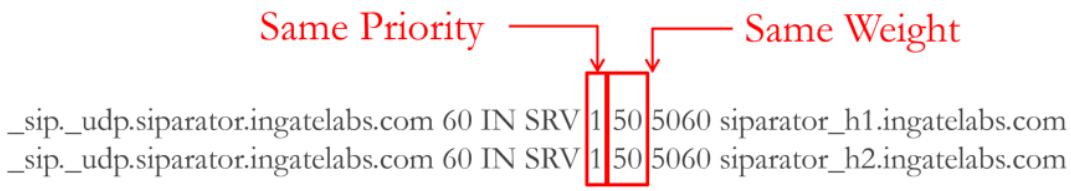
*_sip._udp.**siparator.ingatelabs.com** 60 IN SRV 1 50 5060 siparator_h1.ingatelabs.com*
*_sip._udp.**siparator.ingatelabs.com** 60 IN SRV 2 50 5060 siparator_h2.ingatelabs.com*

4.3.1 Load Balancing

All SRV Records with the same Priority form a load balancing group

Weight is used for distribution in terms of what proportion of traffic will be sent to each destination

For instance, an even distribution between 2 servers might look like this:

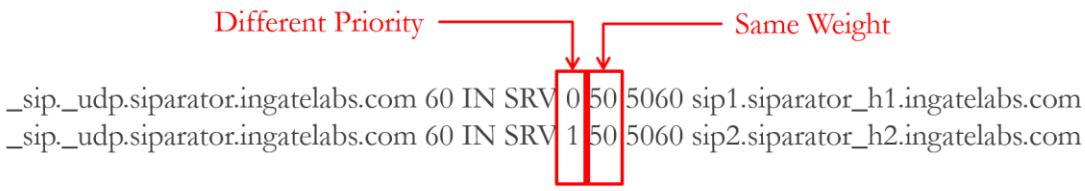


4.3.2 Failover

SRV records with a lower ordinal priority are always tried first

Records with higher ordinal priorities are only tried if all records with lower ordinal priority are tried considered unreachable

For instance, failover between 2 servers might look like this:



In this case, all sip requests to siparator_h1 are always tried first, and only when it is unavailable, to siparator_h2

5 Configuring DNS SRV on AWS

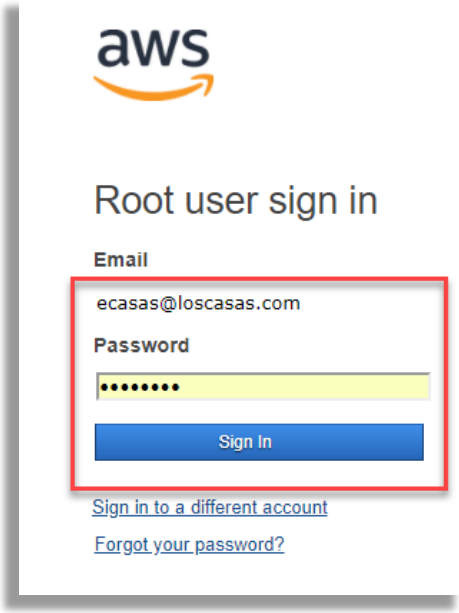
AWS provides its own DNS Services known as ***"Route 53"***.

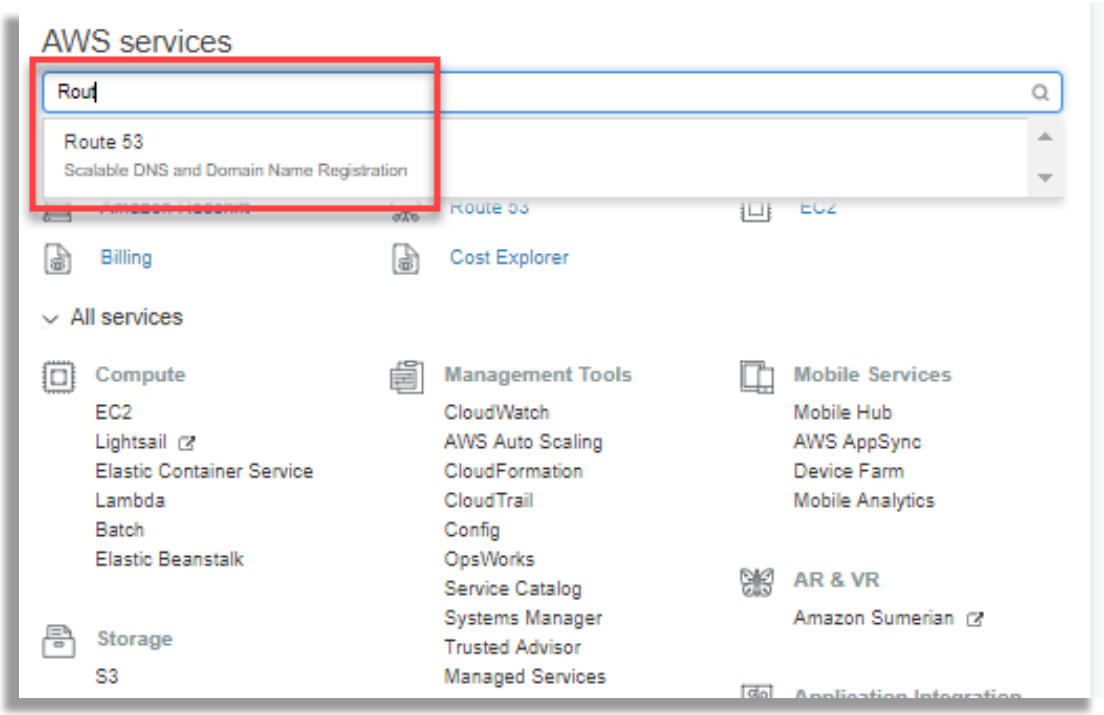
Amazon Route 53 (Route 53) is a scalable and highly available Domain Name System (DNS). It is part of Amazon.com's cloud computing platform, Amazon Web Services (AWS). The name is a reference to TCP or UDP port 53, where DNS server requests are addressed. Route 53's servers are distributed throughout the world.

5.1 Configuring DNS SRV for Public Hosted Zones

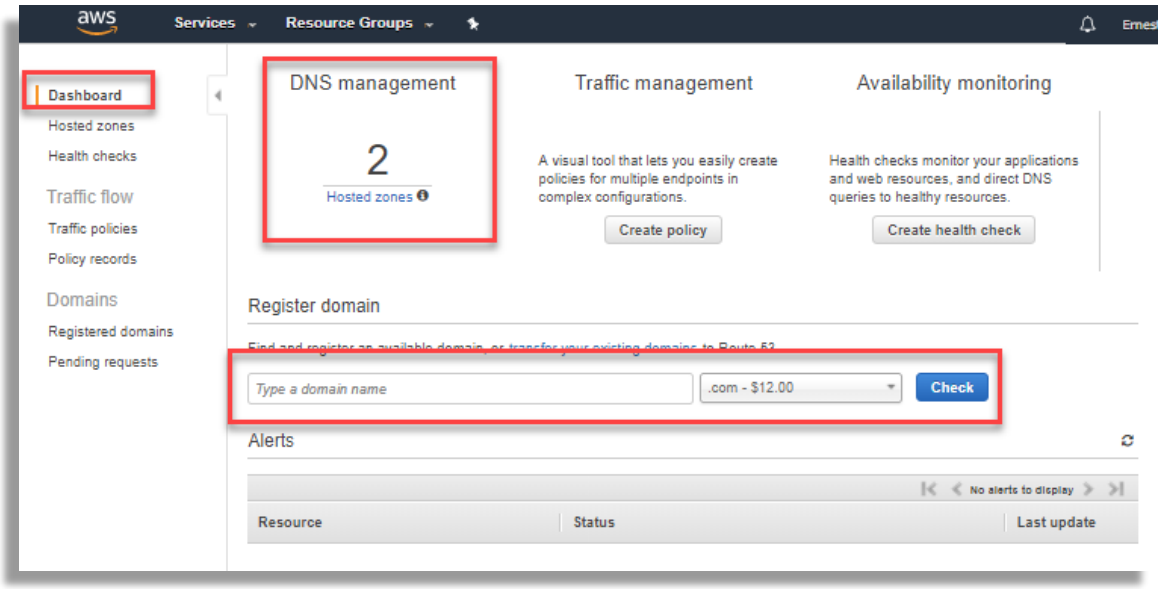
Here we will show all you need to do to create your DNS SRV records in Route53, including the initial setup of your domain.

You need of course, to have an AWS account and you will need to log in to your AWS Console. (http://console.aws.com)





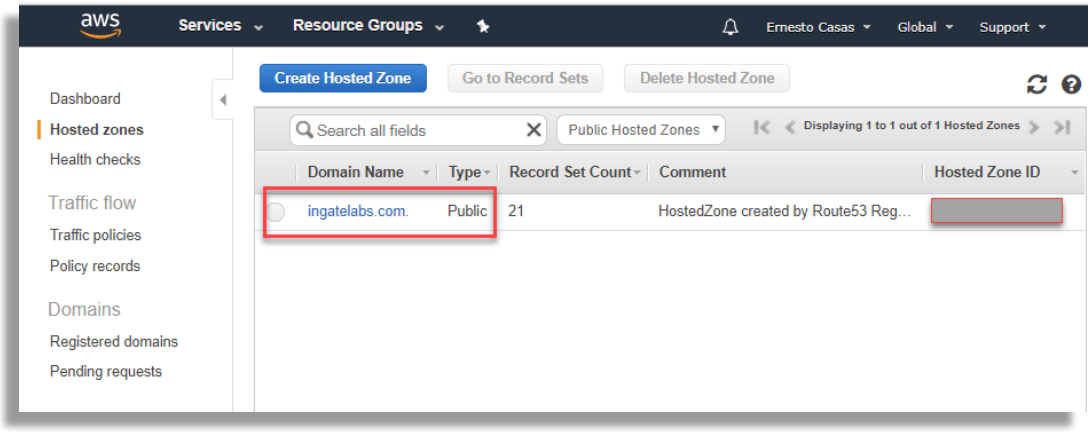
Route 53 dashboard should look like this:



1) Among other options, the dashboard gives you access to manage the service and hosted zones
2) Allows you to create and own a domain in case you need it. This has a cost associated depending on the domain.

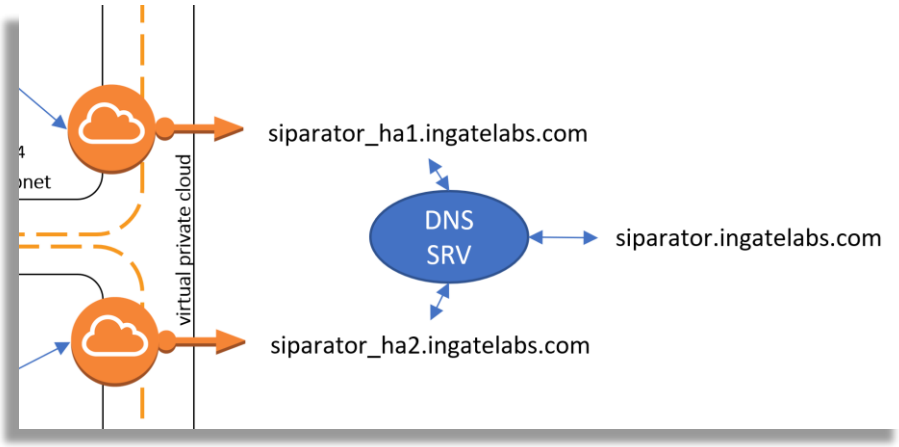
A hosted zone is a collection of records for a specified domain. You create a hosted zone for a domain (such as example.com), and then create records to tell the Domain Name System how you want traffic to be routed for that domain.

In our example we have already created and own a public domain "ingatelabs.com"

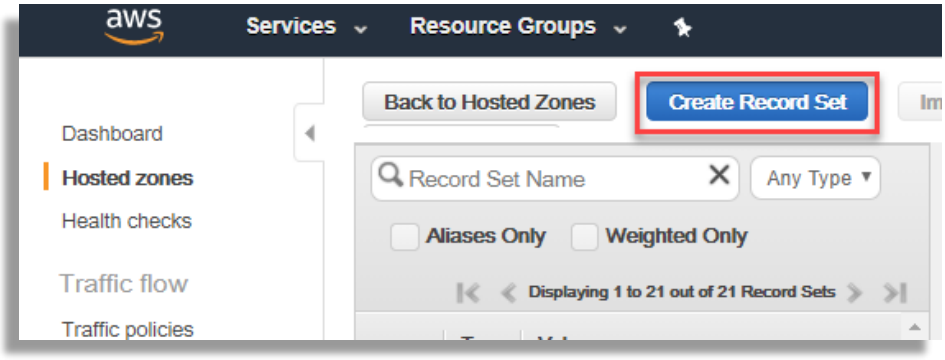


Based on what we have discussed in previous sections we will define all A and SRV records needed to build a failover FQDN using DNS SRV

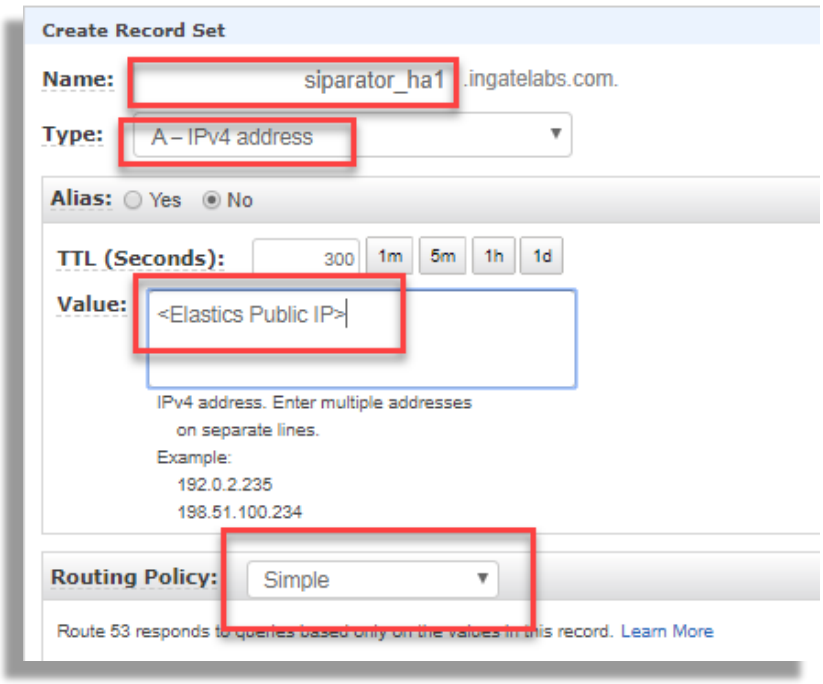
Based in our diagram:



First, we will create an A record for each SIParator and associate them to the corresponding Elastic IP Address

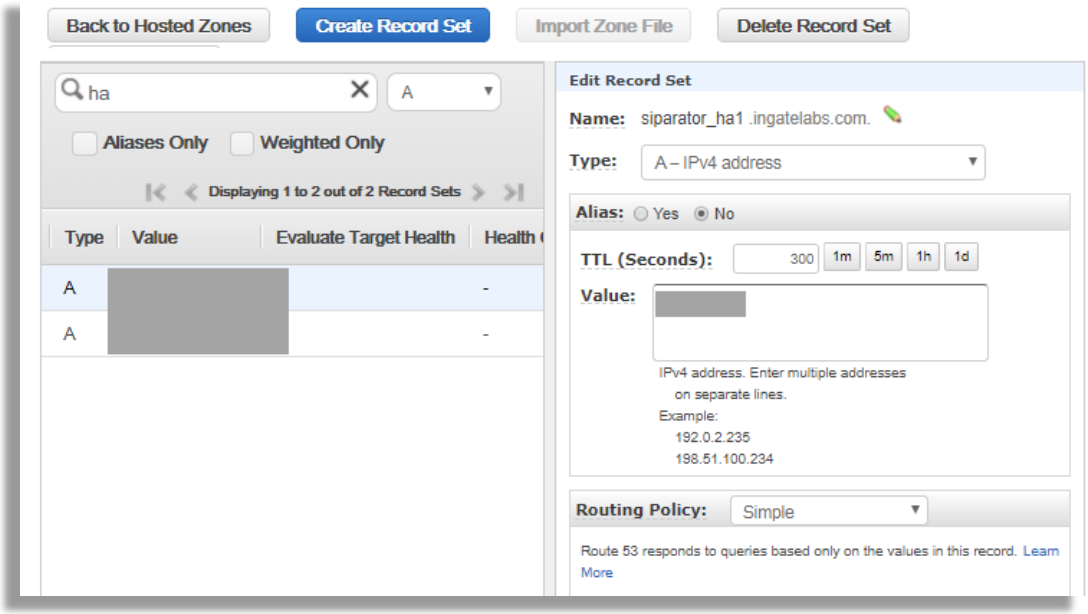


Select the hosted zone and press on "Create Record Set"



1) Create the record selecting Type A-IPv4 Address
2) Enter the Elastic IP address in the Value Field
3) Keep the default Policy on Simple.
4) Press "Create"
5) Repeat the process for siparator_ha2.ingatelabs.com

Once created you can verify everything is as expected:



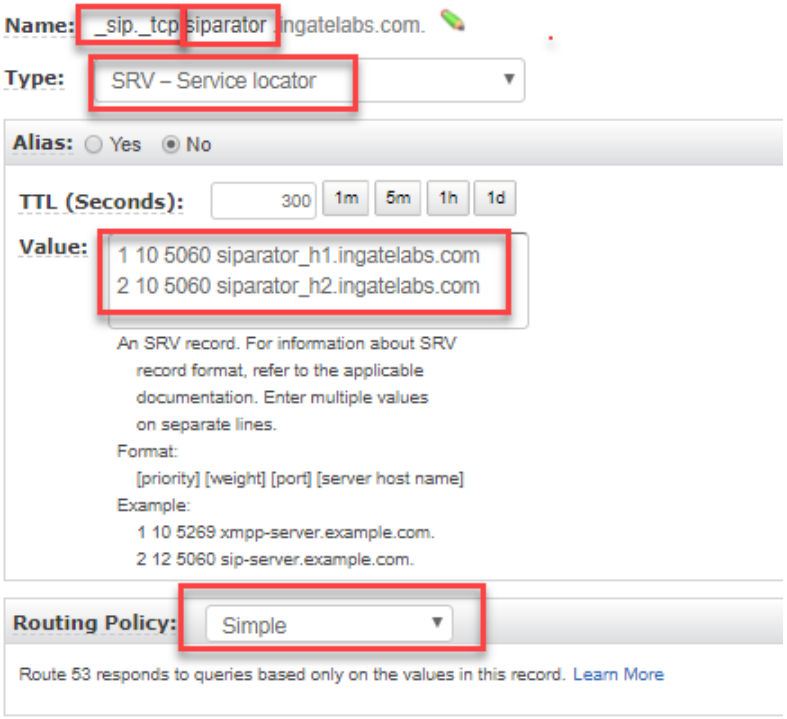
Now we have two new FQDN, each one resolving to one of the corresponding public IP addresses.

Now we will create SRV records for SIP UDP, SIP TCP and SIP TLS.

Three SRV records will be needed (One per transport protocol, UDP, TCP and TLS)

As we want to create a Failover strategy we will use different Priorities and same weights.

Each record should look like this:



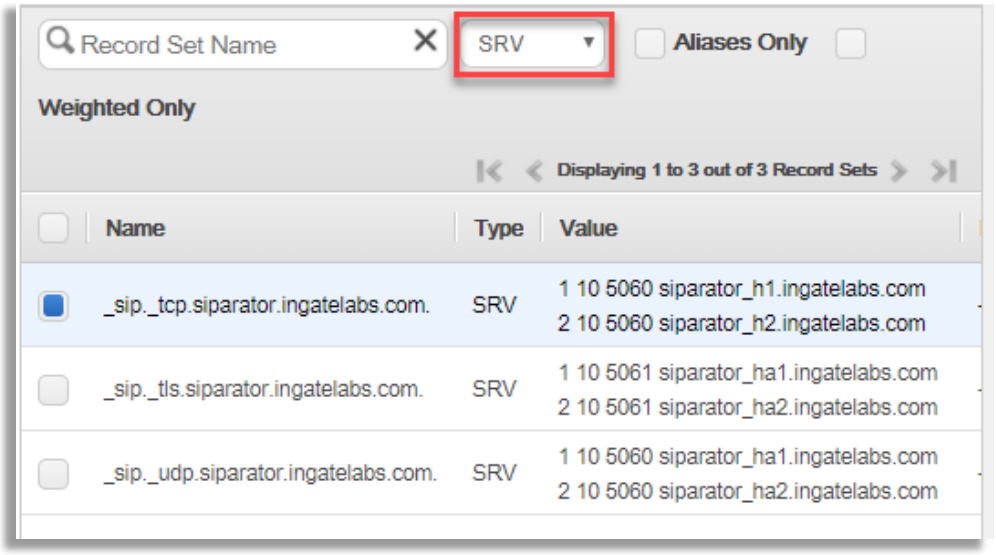
1) Note that Protocol and transport are part of the domain prefix (_sip._tcp),
2) The FQDN that will be referred to resolve is siparator.ingatelabs.com
3) Siparator_h1 will be the primary for failover (lowest priority)
4) _h1 and _h2 have same weight
5) Routing Policy will be kept by default.

You will need to repeat this for UDP (port 5060) and TLS (port 5061) to complete our 3 SRV records.

In Route 53 an SRV record Value element consists of four space-separated values. The first three values are decimal numbers representing priority, weight, and port. The fourth value is a domain name.

Note, other typical parameters as explained in 4.3 are included as part of the domain name (i.e. _sip._udp.siparator), or the TTL parameter that can be setup in the same screen.

As a summary you can verify the SRV records you recently created and should look like these:

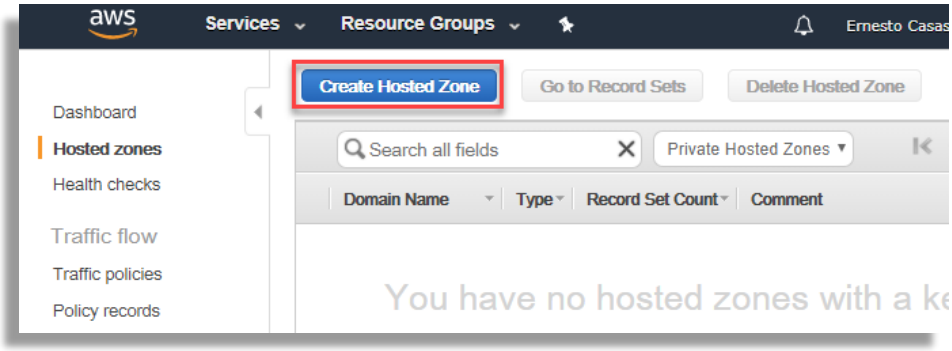


This concludes what needs to be done to create your SRV records with Failover strategy.

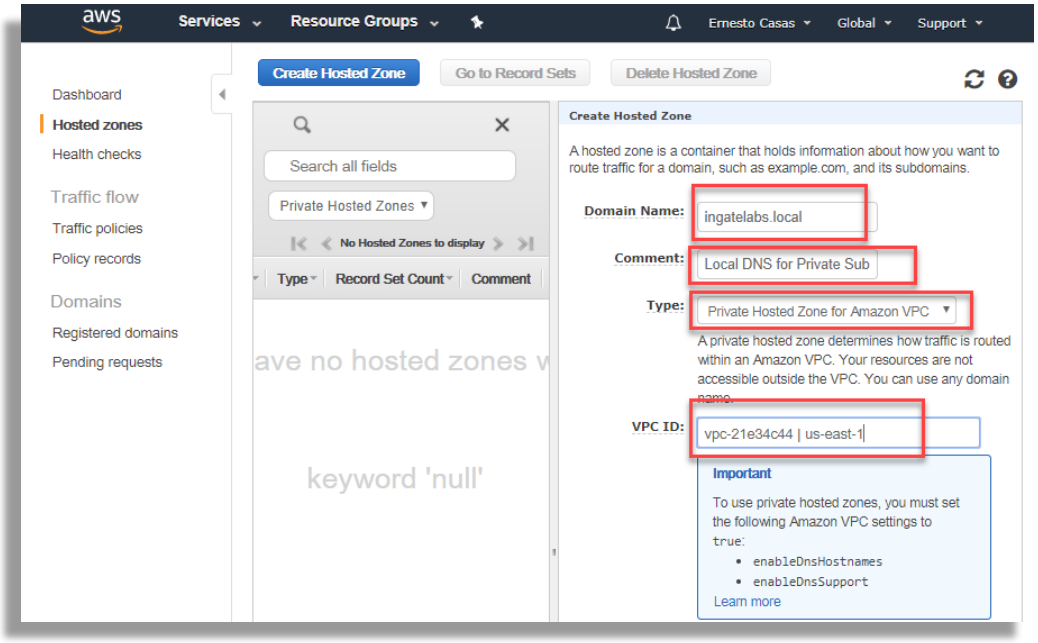
5.2 DNS SRV as an additional option for Failover in the private subnet

Later we will show 2 methods to implement Failover for the PBX. One of them is also based on SRV records but used to resolve IPs in the private network. To do so we will create a private hosted zone for the private network using a private domain named "ingatelabs.local".

1) First, we need to create a private Hosted Zone



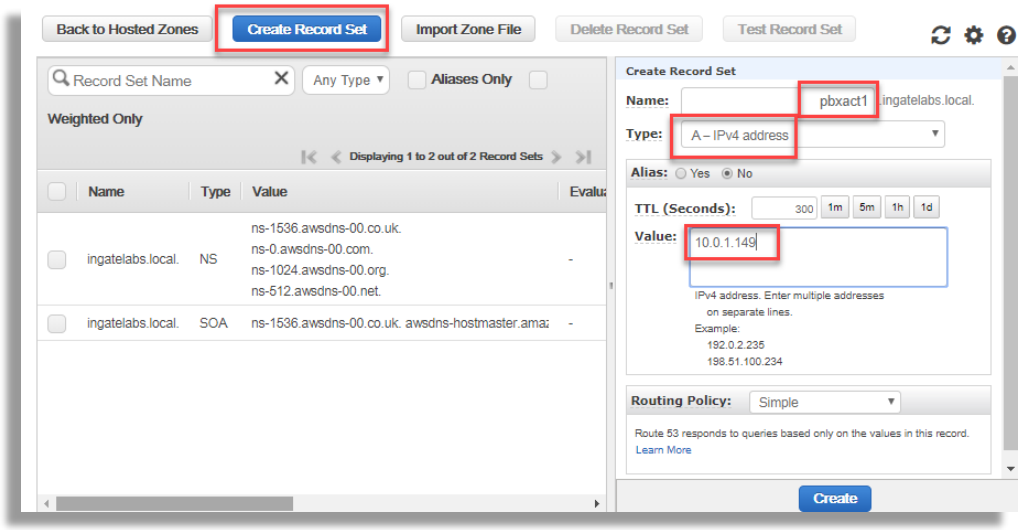
2) Assign the Domain Name (ingatelabs.local), Select Private Hosted Zone for VPC, and press "create"

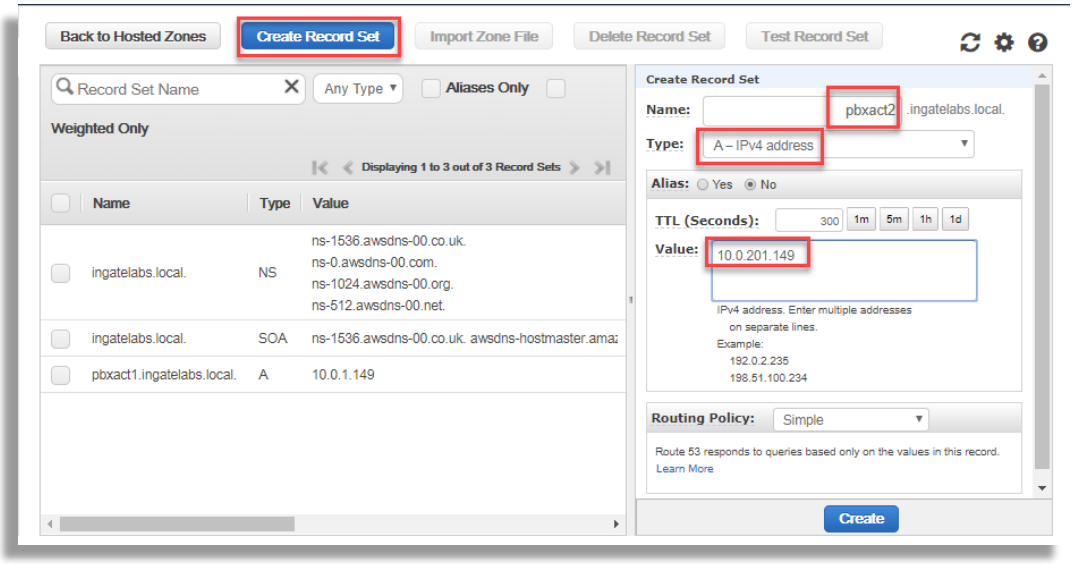


Make sure you associate this domain to the VPC you are using to deploy the solution

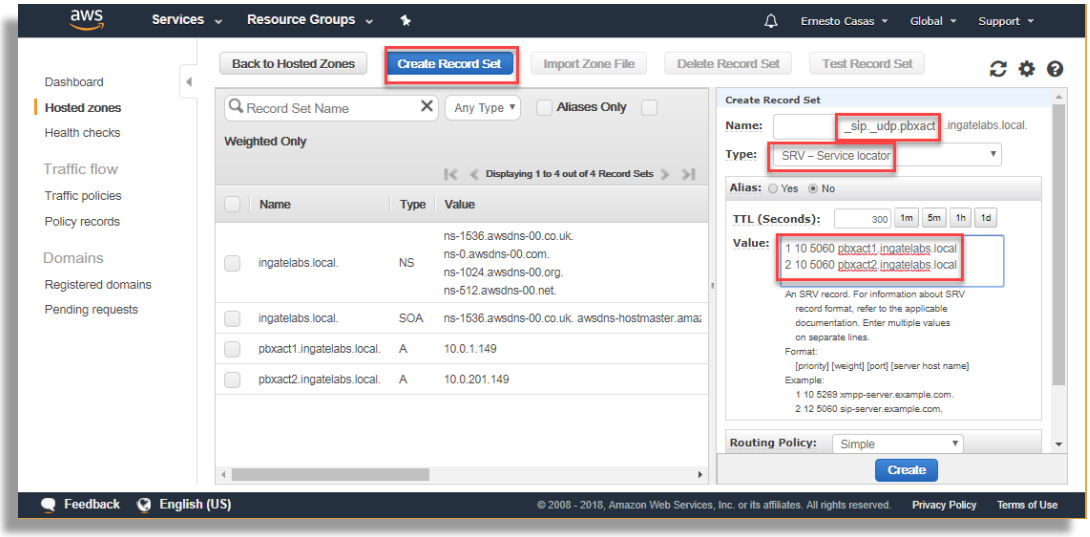
You can add more VPCs as needed in case your SIParators or PBXs are installed in more than one VPC. For more detailed information you can refer to: https://docs.aws.amazon.com/Route53/latest/DeveloperGuide/hosted-zones-private.html

3) Create A records for each PBX. Based on the diagram they are located at 10.0.1.149 and 10.0.201.149.





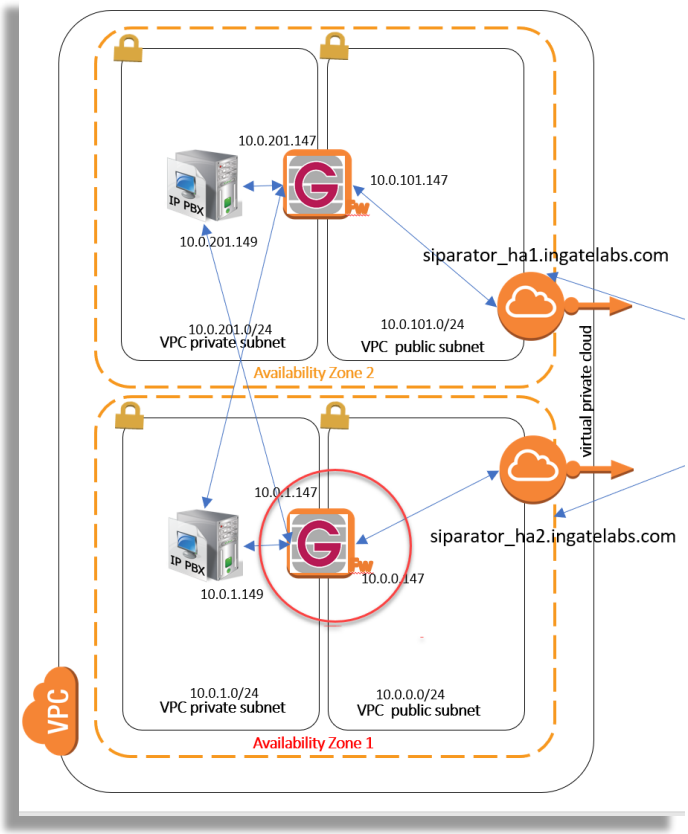
4) Create SRV Record for Failover strategy between the two PBXes



You can repeat this step if you need to add TCP and TLS as additional transport options.

5.3 SIParator configuration for Failover using DNS SRV and other features (SIP Trunking use case)

First, we need to understand that we will setup 2 SIParators with almost identical configuration. Just a few elements will be different between them which we explain in detail later.

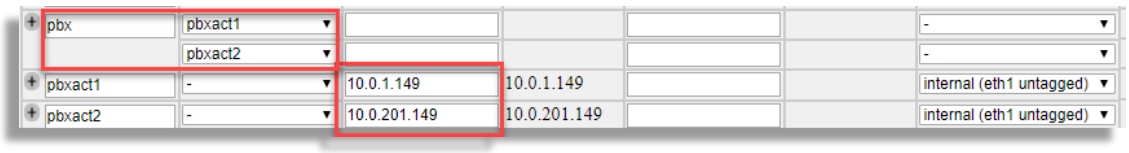


Let's first focus on configuring one of the SIParators

5.3.1 IP-PBX Failover

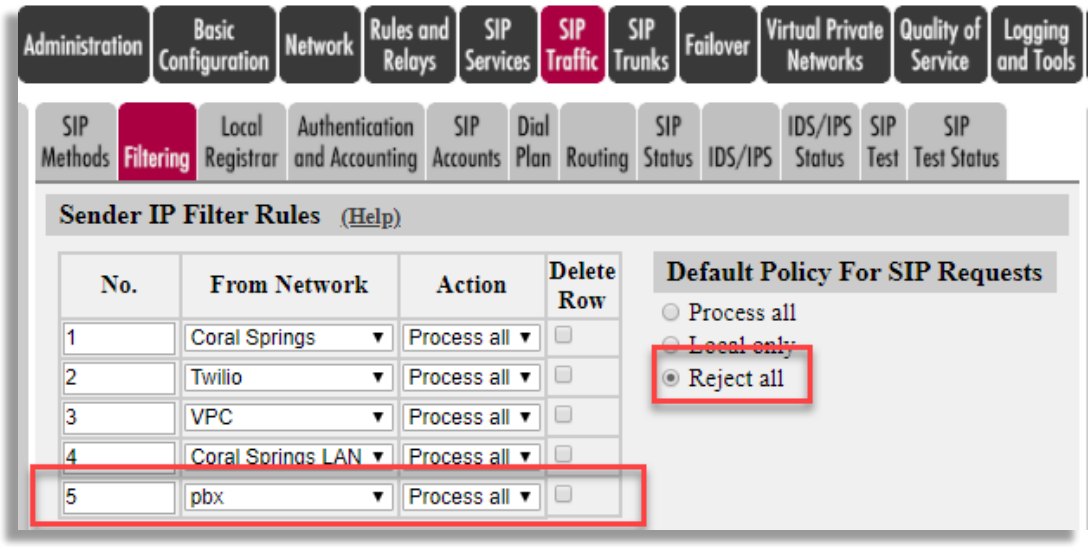
We are not going to present the details of all the required configuration but will describe the ones that are related to some type of failover functionality.

1) As shown in the diagram we will provide failover inbound calls to 2 IP-PBXes, and, we will allow traffic coming from any of them. So, we will need in the "Networks and Computers" Section under "Network":



Here with a single name "pbx" we are including any of the two PBXes shown in the diagram.

2) In "Filtering" under SIP Traffic we are adding "pbx" as a valid source to process SIP requests

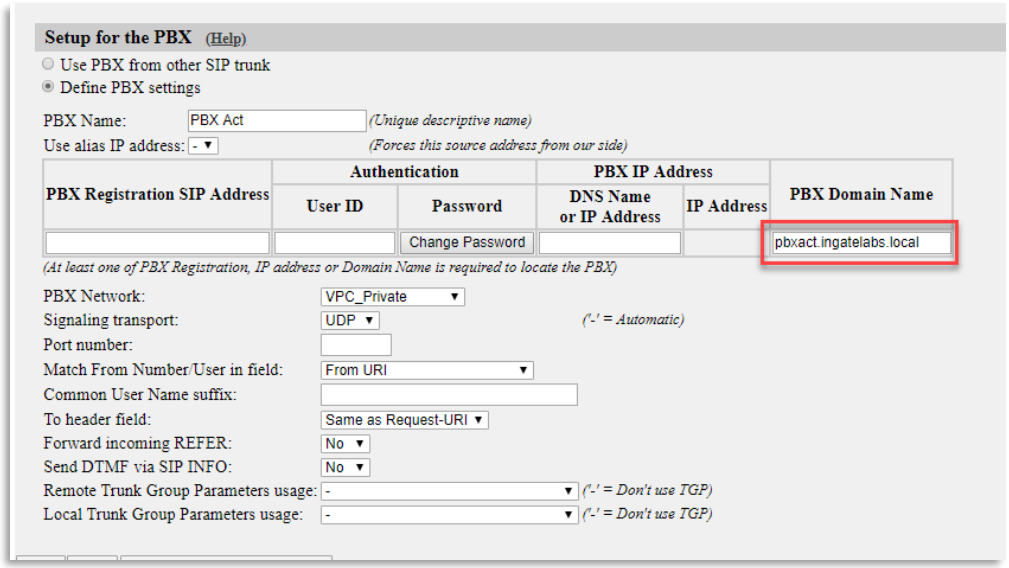


Here, as we are using "pbx" network name, the 2 IP-PBXes has been included as a valid source.

3) Inbound failover for PSTN calls. In this case, we configure how to failover to a secondary pbx (pbxact2) in case the primary does not respond to inbound calls coming from PSTN.

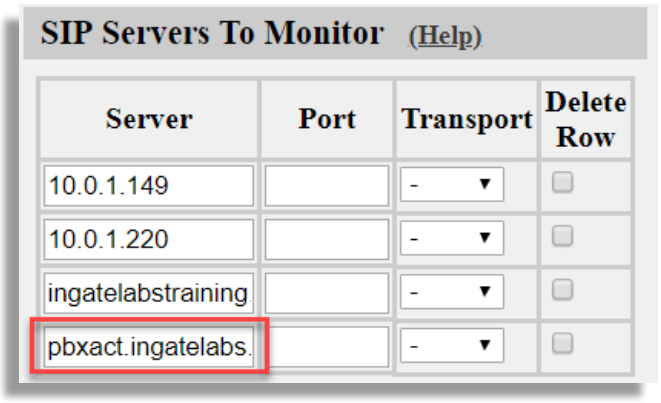
   There are 2 methods we can use, the first one is based on DNS SRV Records for Private Hosted Zone (See section 5.2). In this case we will use pbxact.ingatelabs.local as the PBX domain.

5.3.2 PBX Failover using DNS SRV for SIP Trunks

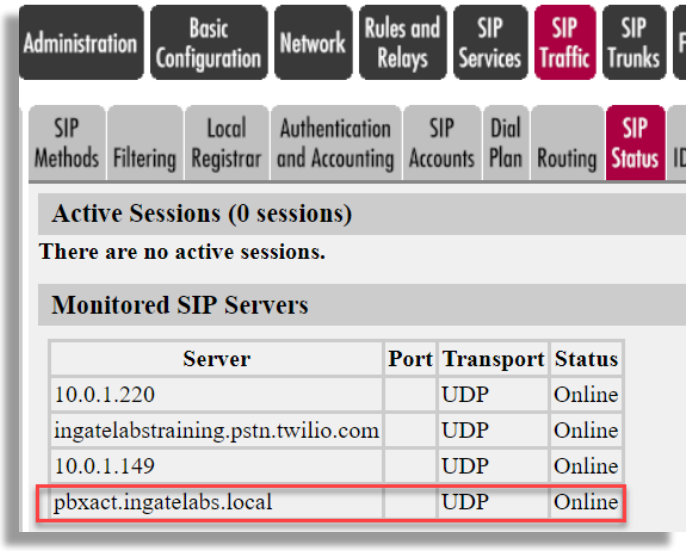


As shown, we are using the DNS name as the PBX Domain associates to an SRV Record. Failover between the two PBXes will be managed by DNS Services based on the Strategy created in Section 5.2

We will add also monitoring to the PBX Domain (pbxact.ingatelabs.local) in the SIP Services Section, Basic Configuration:



You can also confirm that the domain is properly resolved using DNS SRV and the PBX is declared as Online. Looking at the "SIP Traffic" section, "SIP Status"

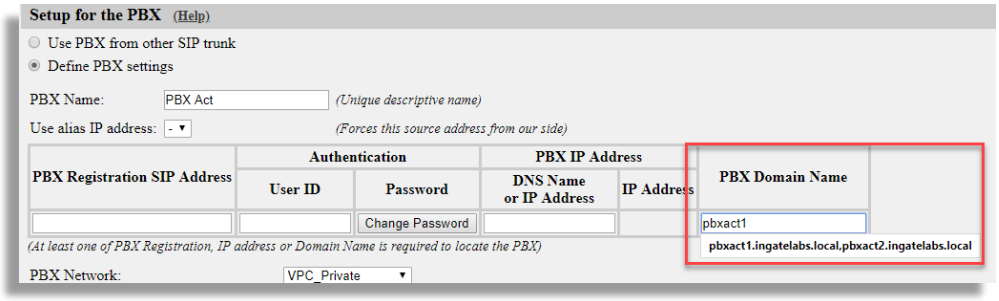


5.3.3 PBX Failover using Dual Homing for PBX Domain in the Trunk Group

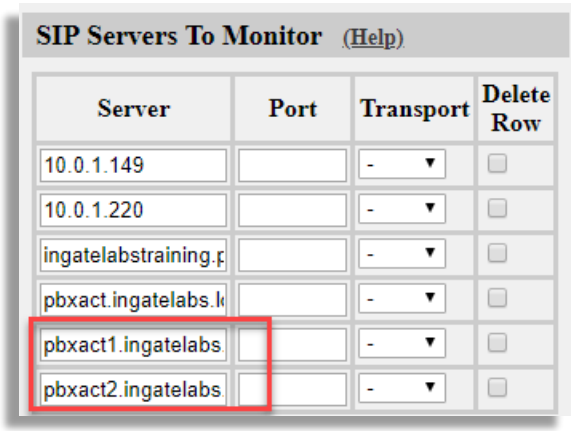
As a second option, we can take advantage of dual domain (two domains comma-separated in the trunk Group PBX configuration). In this case, we can directly use PBX IP addresses or domain names created with A Records as explained in section 5.2.

PBX Domain Name - Optional SIP domain name of the PBX in case the PBX wants incoming calls to be addressed to sip:number@domain instead of sip:number@ip-address. ***You shall also use this field if you have two redundant PBXes. Then add the IP address or FQDN of both PBXes in this field, separated by comma.*** (Do not fill in the PBX IP Address field then.) If the first PBX is out of service, the second will be tried. You should also enter these

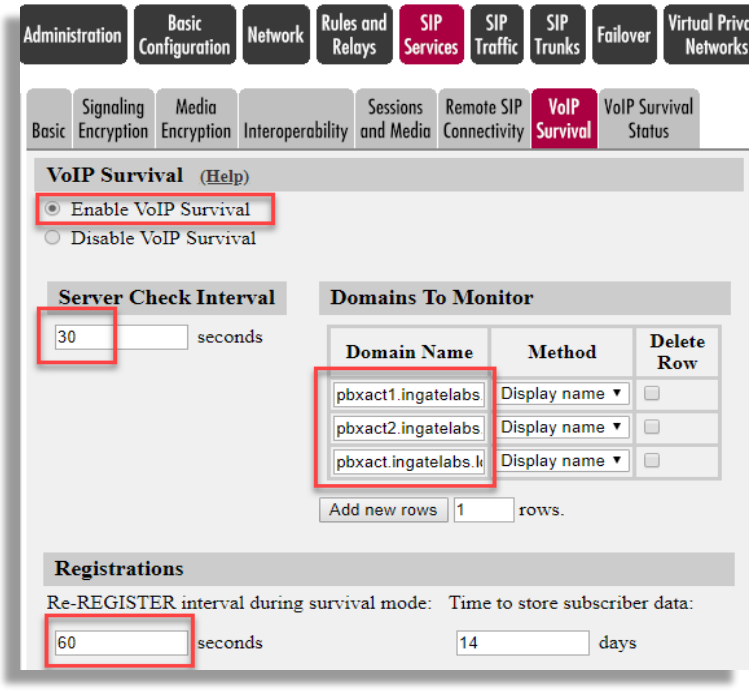
PBXes for monitoring at the SIP Services > VoIP Survival page to speed up failover.



Adding the two domains to SIP Monitor:

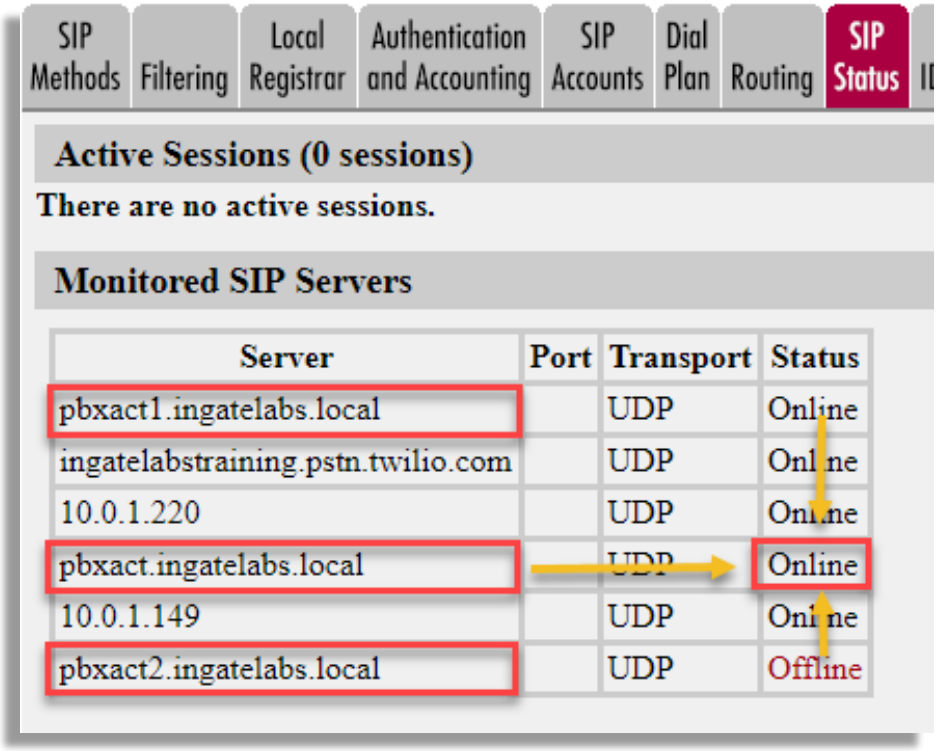


And, we can speed up the Failure detection by enabling the VoIP Survival feature and adding both domains (A Records). We can also include an SRV record to illustrate that it also works for the DNS SRV case.

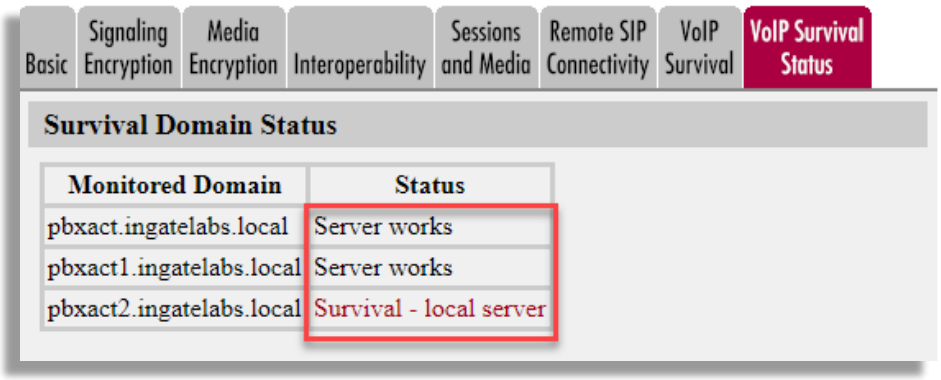


Note the field Server Check Interval. This value in seconds defines how many seconds a new check (SIP OPTIONS) will be sent. The interval must be shorter than the SIP blacklist interval on the Sessions and Media page.

Here is how the SIP Status screen will look, in this case, one of the two IPPBX shows as down, and having at least one available shows the SRV Domain as online.



As we are using VoIP Survival, the same behavior is expected:



5.3.4 SIP Trunks Failover.

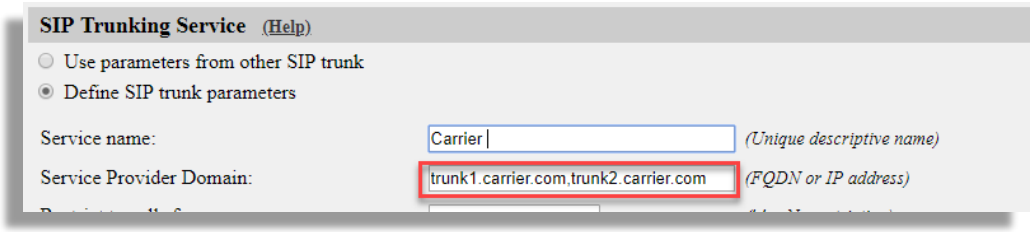
PSTN connectivity is another of the important components of the service to consider when designing a highly available service. In our case, we will mention the most common techniques to provide a failover strategy. In some cases, to have a

highly available solution for PSTN connections depends on how the ITSP offers their service.

1) **Dual/Multi Homed Carrier**. In this case the ITSP provides more than one IP Address or FQDN and the SBC should be able to allocate them associated to the same trunk
2) **DNS SRV SIP Trunk (Carrier Side)**. In this case the carrier will provide an FQDN that resolves using SRV records pointing to more than one destination. The resilience strategy is managed using DNS SRV, including Failover and Load Balance Strategies
3) **DNS SRV SIP Trunk (SIParator Side).** The other side of the coin could be to provide the carrier an FQDN for inbound traffic, and such an FQDN is resolved using DNS SRV Records. Is short, the ITSP must support DNS SRV resolution for customer trunk destinations
4) **Multi IP Hunting (Carrier side).** In this case, The ITSP must be able to hunt, failover or load balance between the IPs or FQDN (A records) pointing to the SIParators
5) **Twilio Redundancy options.** Twilio is a good example of an ITSP which provides more than one option for resilience. First, they provide each customer SIP trunk a unique FQDN with SRV records, solving the outbound failover case. Second, they provide DNS SRV support for the customer to provide an FQDN with SRV records for inbound failover. They also provide inbound traffic a way to assign multiple FQDNs or IP addresses to send inbound traffic using a Failover or balancing strategy.

5.3.5 Dual/Multi homed Carrier.

In this case, we will take advantage of Dual Domains supported in SIP Trunk configuration under the Trunk Group Pages

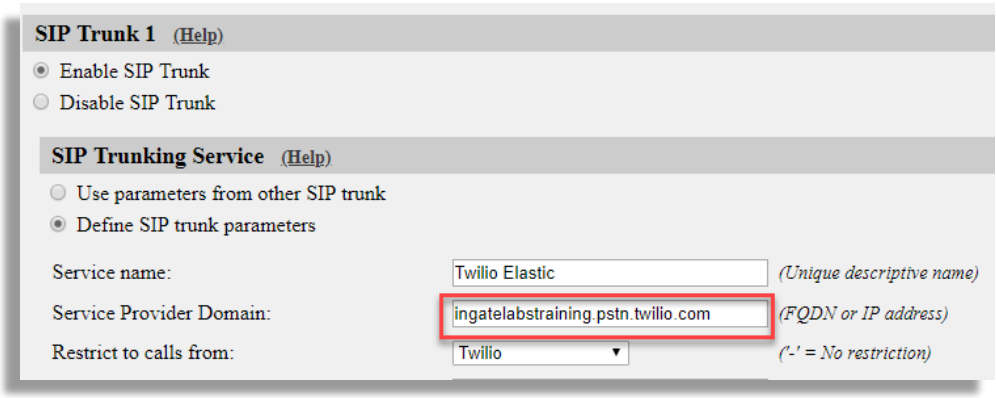


Service Provider Domain - The FQDN or IP address of the ITSP SIP server. This domain name will be used in the Request-URI and To header field for outgoing SIP requests. ***If there are two redundant SIP Servers, enter both here, separated by comma. (Do not enter both SIP Servers if these instead are addressed by DNS SRV records for the Outbound Proxy.)*** If the first SIP Server is out of service, the second will be tried. You should also add these SIP Servers to the table "SIP Services" > Basic > "SIP Servers to Monitor" to accelerate failover.

5.3.6 DNS SRV SIP Trunk (Carrier Side)

In this case the carrier provides a SIP Trunk Domain (FQDN) with DNS SRV Records. This helps to have a failover strategy on outbound traffic to PSTN

This is the case of Twilio and other advanced SIP Trunk Providers.



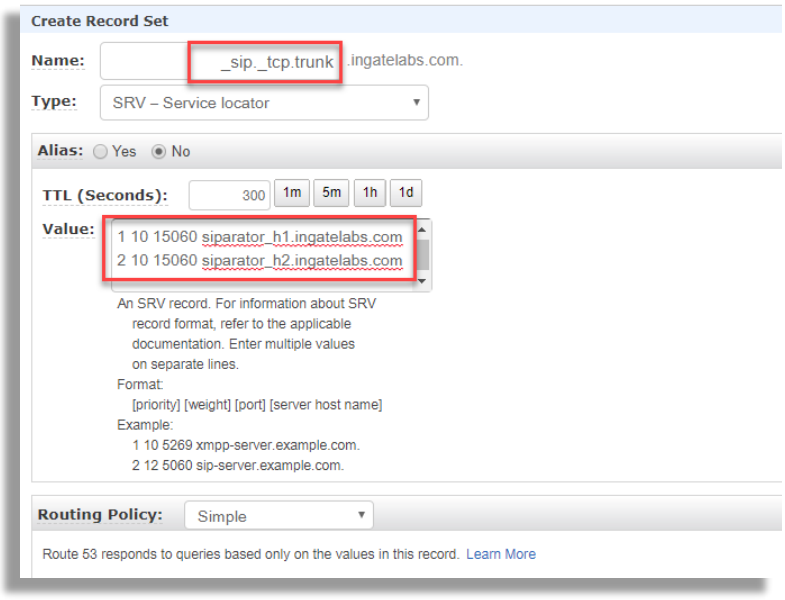
Notice the domain used is an FQDN resolved using SRV records, where the carrier Failover strategy will be implemented.

5.3.7 DNS SRV SIP Trunk (SIParator Side).

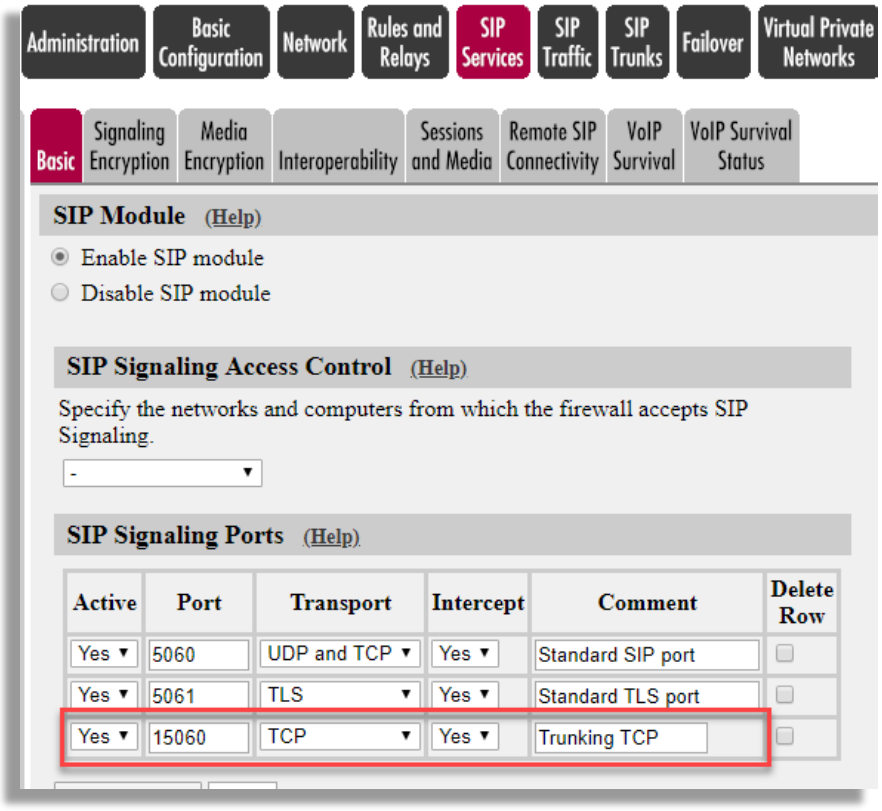
In this case, as explained in section 5.1, the SIParator is reachable via an FQDN using SRV records. We can reuse the same siparator.ingatelabs.com or we can create additional FQDN specifically for SIP Trunking.

For instance, let's think how to implement a specific trunk using a new FQDN (trunk.ingatelabs.com) and let's use port 15060 and TCP to failover to siparator_ha1.ingatelabs.com and siparator_ha2.ingatelabs.com

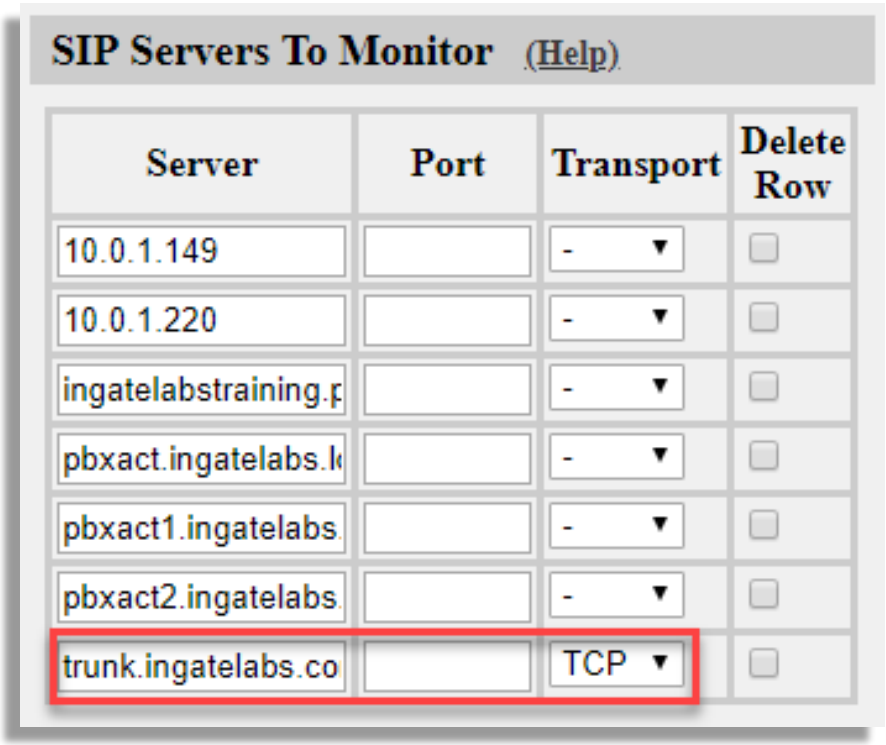
1) Create a Record Set in the Hosted Public Zone ingatelabs.com



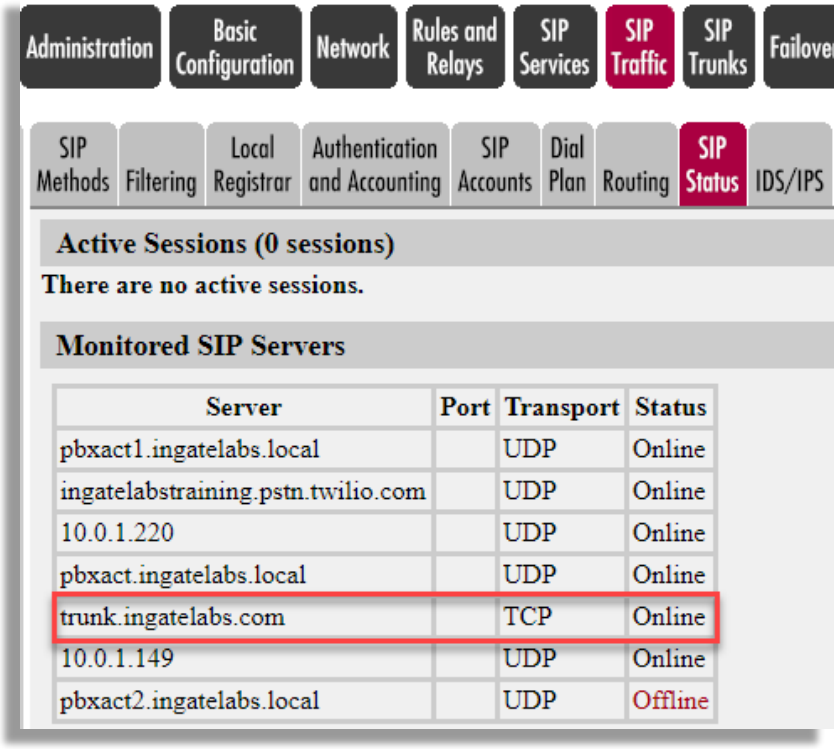
2) Add 15060 on TCP as a valid SIP listening port



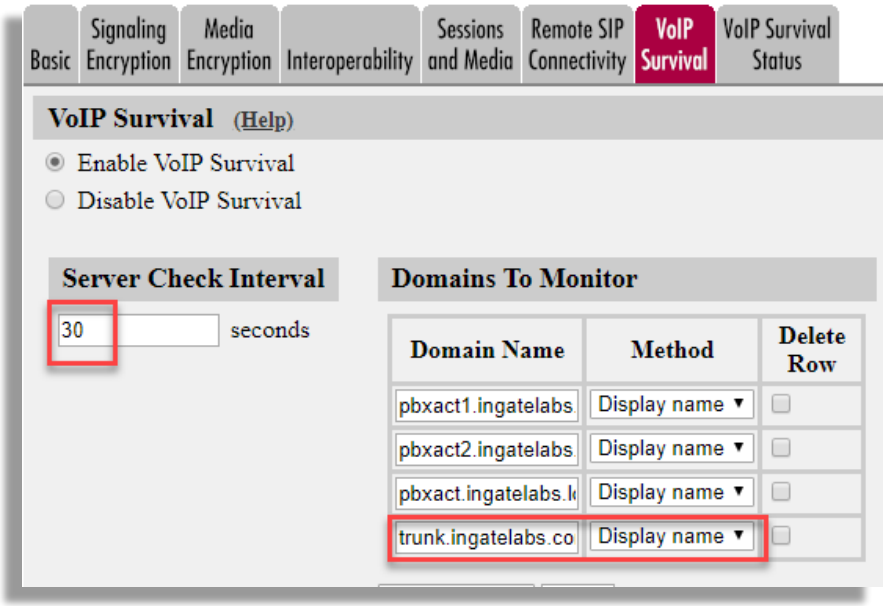
3) Add SIP Monitor for trunk.ingatelabs.com



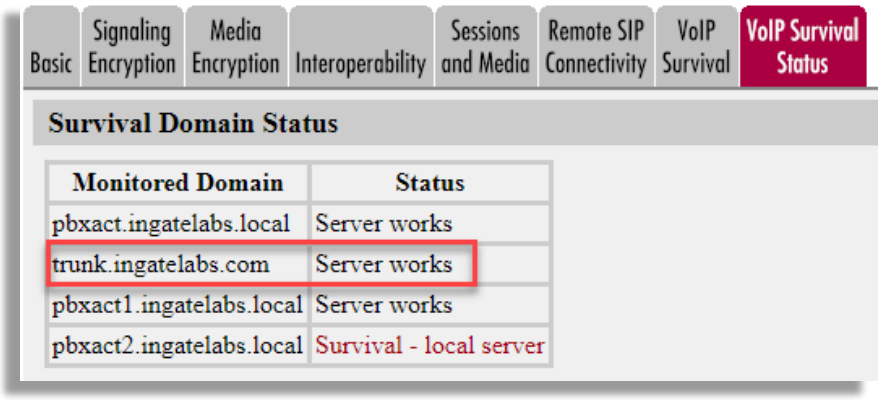
SIP Monitor will show the status for the Trunk Domain



4) Add VoIP Survival for trunk.ingatelabs.com



VoIP Survival Status will show:



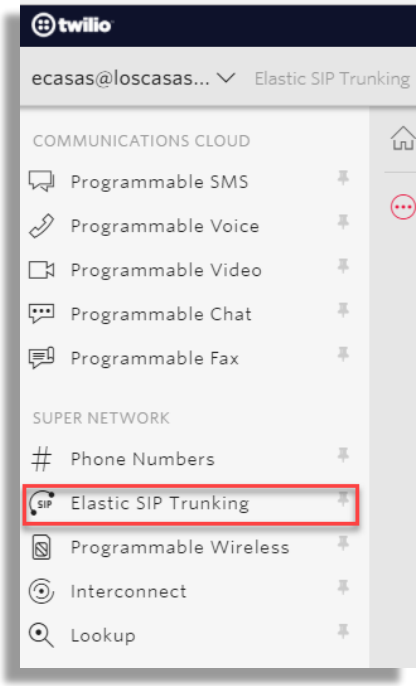
5.3.8 Twilio Redundancy Options

Twilio, among several SIP and SMS APIs and Services, also provides very advanced SIP Trunking Options.

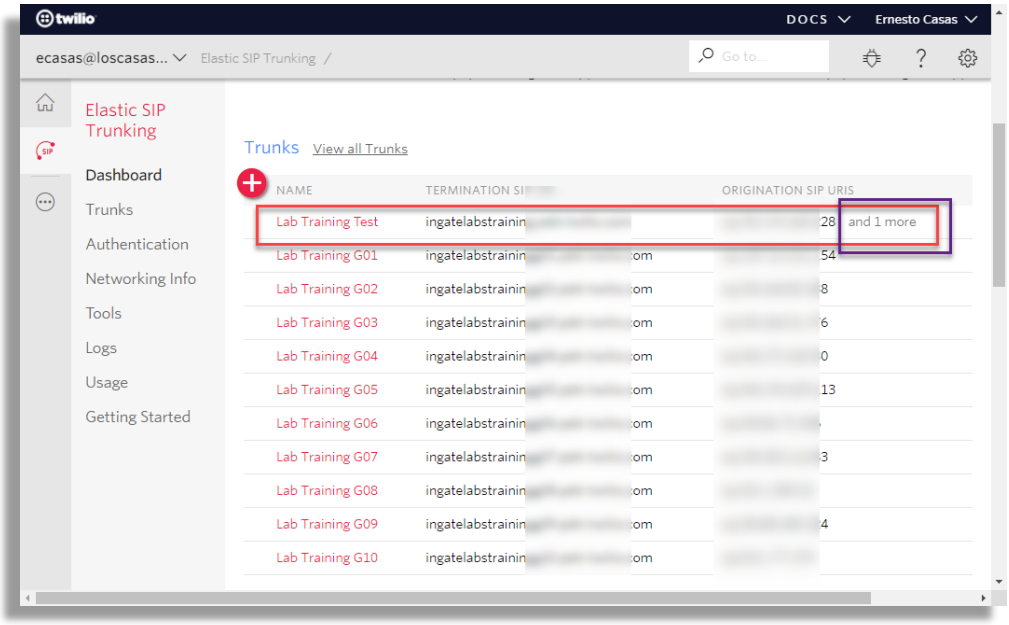
When designing a Highly Available solution, it is very important to review not only features and capabilities provided by your own infrastructure but also related functionalities on the Service Provider side.

In this section we will use Twilio as the Service Provider for their Failover capabilities. Many other ITSPs in the market offer extensive HA features and you should consider them when designing your platform.

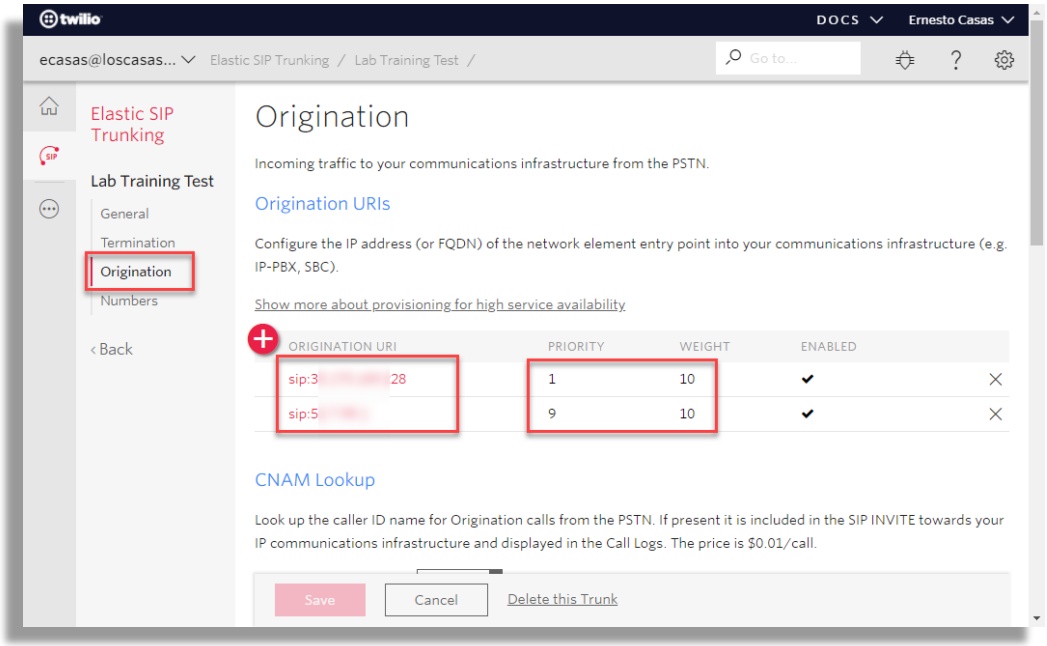
Lets first access Twilio Account Dashboard → Elastic SIP Trunking



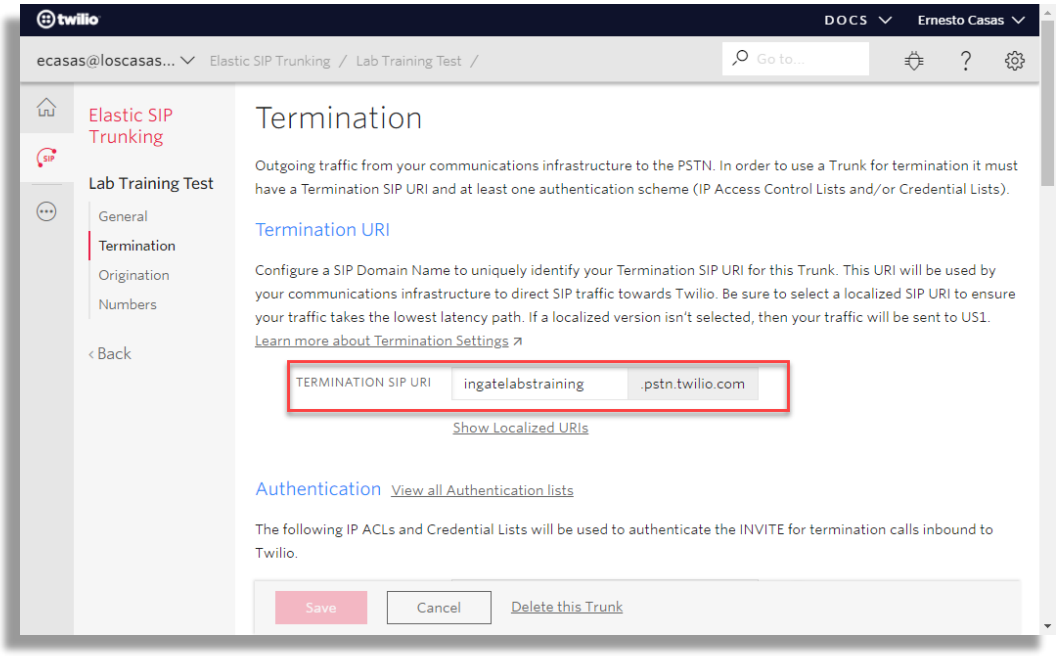
For this document, we set up the Lab Training Test SIP Trunk to be configured in such a way that calls can be routed to any of the 2 SBCs using a failover strategy.



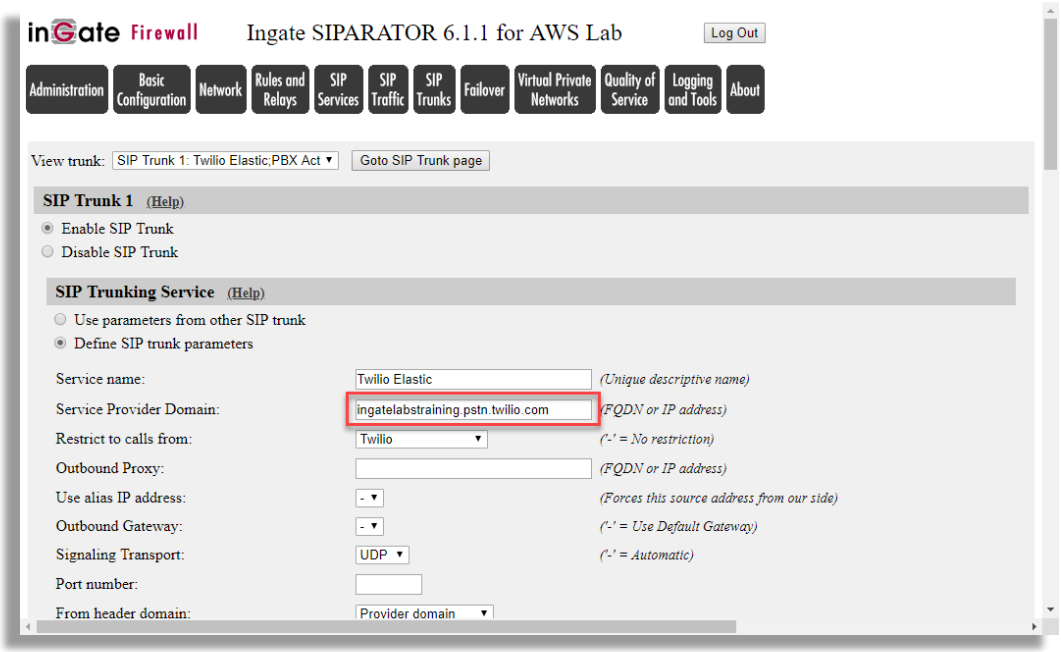
Now we can associate one or more origination IP addresses or FQDN. Notice that when using 2 or more, you have the tools to assign priority and weight and they work exactly how DNS SRV records work. You can have a Failover or even a Load Balance Strategy.



Regarding origination, Twilio will allow you to define a URI for each SIP Trunk that you will use in your SIParator Trunk Group configuration.

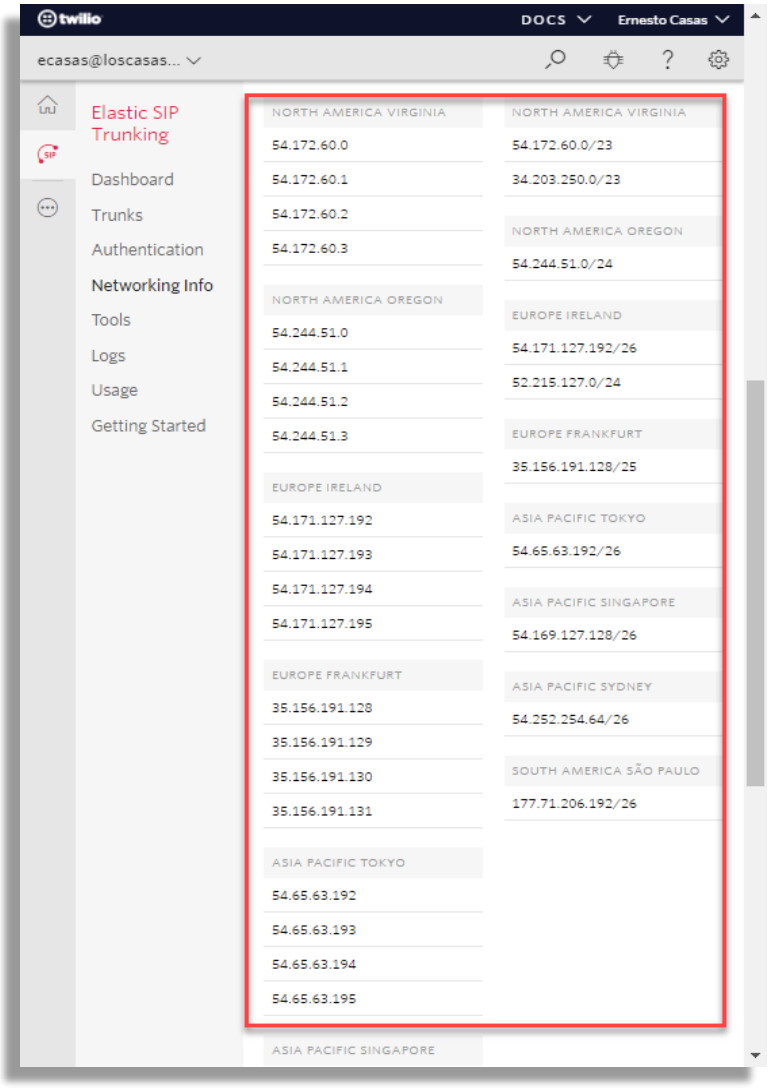


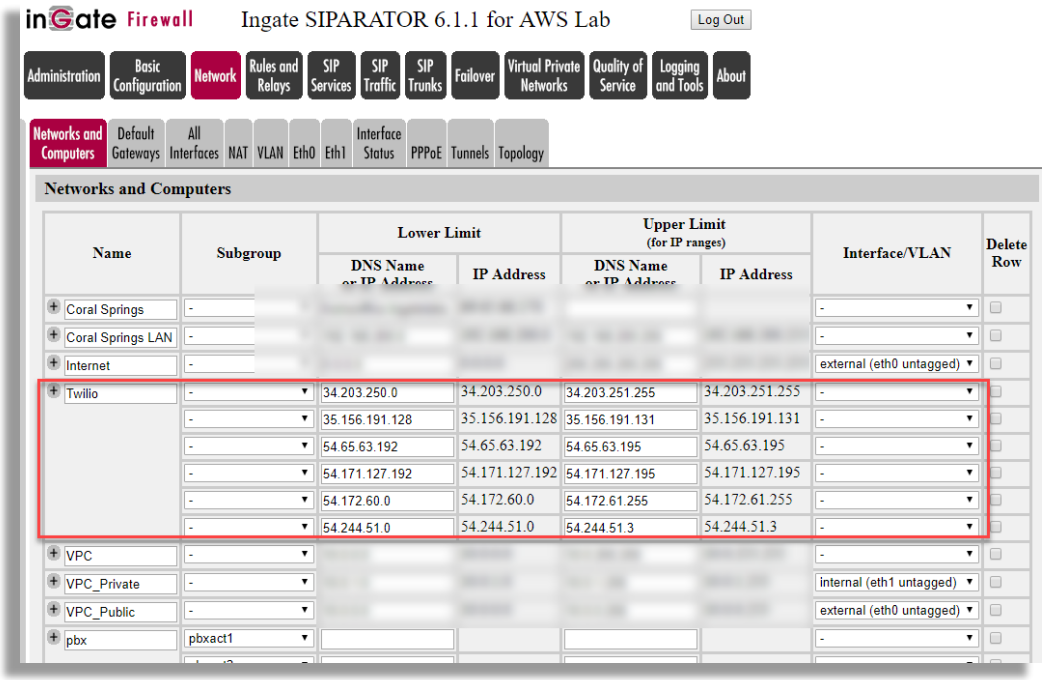
The URI defined above "ingatelabs.pstn.twilio.com" will be the one used in SIParator Trunk Group Page configuration, as shown here:



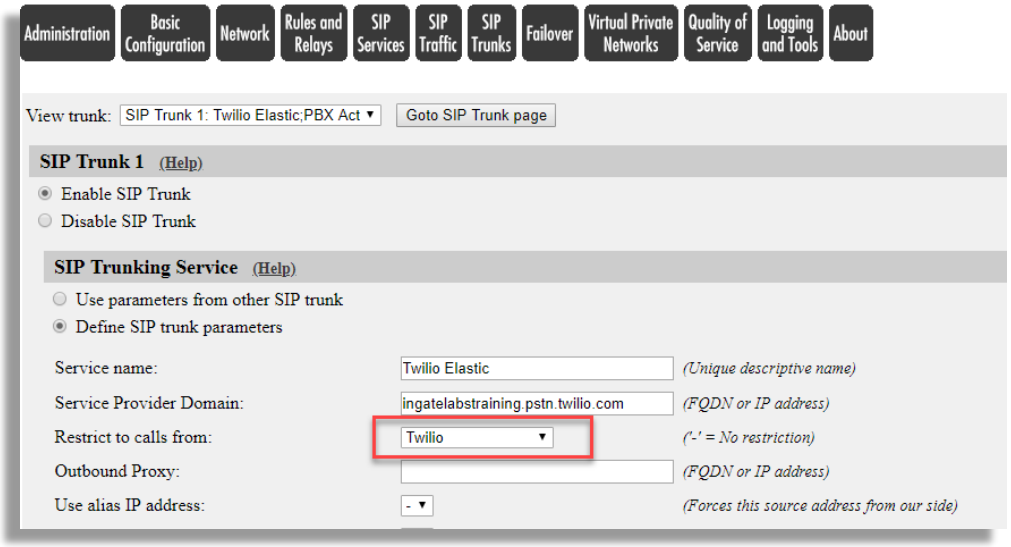
The domain ingatelabstraining.pstn.twilio.com resolves to multiple addresses predefined by Twilio and they do what's needed to ensure availability of any of them. Those IPs are listed per region in your dashboard → Elastics Sip trunks → Networking Information and are used in a round-robin fashion in the region you have your Instances deployed.

These IP addresses should be added to your Networks & Computers section in SIParator configuration

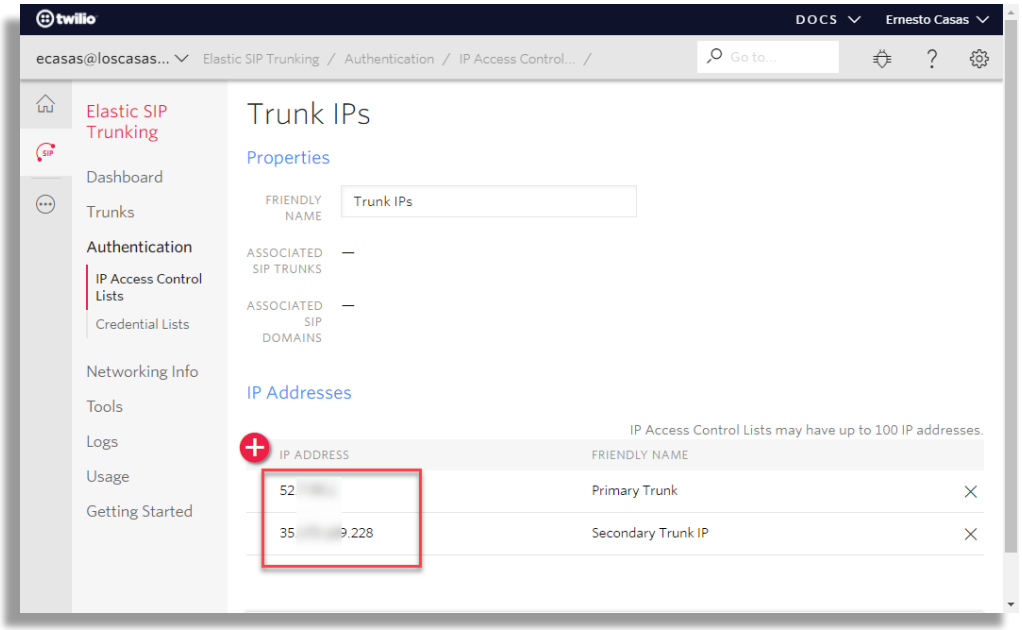

twilio
DOCS
Ernesto Casas
ecasas@loscasas...
Elastic SIP Trunking
Dashboard
Trunks
Authentication
Networking Info
Tools
Logs
Usage
Getting Started
NORTH AMERICA VIRGINIA
54.172.60.0
54.172.60.1
54.172.60.2
54.172.60.3
NORTH AMERICA OREGON
54.244.51.0
54.244.51.1
54.244.51.2
54.244.51.3
EUROPE IRELAND
54.171.127.192
54.171.127.193
54.171.127.194
54.171.127.195
EUROPE FRANKFURT
35.156.191.128
35.156.191.129
35.156.191.130
35.156.191.131
ASIA PACIFIC TOKYO
54.65.63.192
54.65.63.193
54.65.63.194
54.65.63.195
ASIA PACIFIC SINGAPORE
NORTH AMERICA VIRGINIA
54.172.60.0/23
34.203.250.0/23
NORTH AMERICA OREGON
54.244.51.0/24
EUROPE IRELAND
54.171.127.192/26
52.215.127.0/24
EUROPE FRANKFURT
35.156.191.128/25
ASIA PACIFIC TOKYO
54.65.63.192/26
ASIA PACIFIC SINGAPORE
54.169.127.128/26
ASIA PACIFIC SYDNEY
54.252.254.64/26
SOUTH AMERICA SÃO PAULO
177.71.206.192/26


inGate Firewall
Ingate SIPARATOR 6.1.1 for AWS Lab
Log Out
Administration
Basic Configuration
Network
Rules and Relays
SIP Services
SIP Traffic
SIP Trunks
Failover
Virtual Private Networks
Quality of Service
Logging and Tools
About
Networks and Computers
Default Gateways
All Interfaces
NAT
VLAN
Eth0
Eth1
Interface Status
PPPoE
Tunnels
Topology
Networks and Computers
Name
Subgroup
Lower Limit
DNS Name or IP Address
IP Address
Upper Limit (for IP ranges)
DNS Name or IP Address
IP Address
Interface/VLAN
Delete Row
Coral Springs
Coral Springs LAN
Internet
external (eth0 untagged)
Twilio
34.203.250.0
34.203.250.0
34.203.251.255
34.203.251.255
35.156.191.128
35.156.191.128
35.156.191.131
35.156.191.131
54.65.63.192
54.65.63.192
54.65.63.195
54.65.63.195
54.171.127.192
54.171.127.192
54.171.127.195
54.171.127.195
54.172.60.0
54.172.60.0
54.172.61.255
54.172.61.255
54.244.51.0
54.244.51.0
54.244.51.3
54.244.51.3
VPC
VPC_Private
internal (eth1 untagged)
VPC_Public
external (eth0 untagged)
pbx
pbxact1

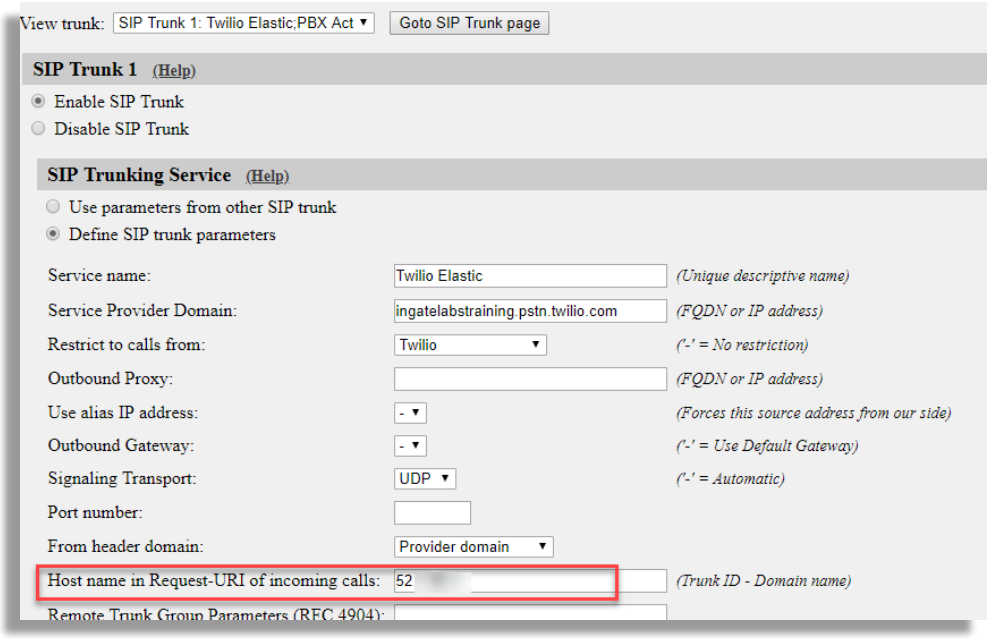
Having Twilio defined as a Network group name will allow you to refer to those addresses by name. For instance, to limit from where SIP trunk traffic is accepted.



It is also required to have an access control list (ACL) to control which IP addresses are included as potential originator of traffic in this trunk from the Twilio point of view. It must be the Public IP addresses assigned to each one of our SIParators.



Finally, Twilio will use those IP address in the host Request URI and it is used to match inbound traffic to the SIParator. It should be included here:

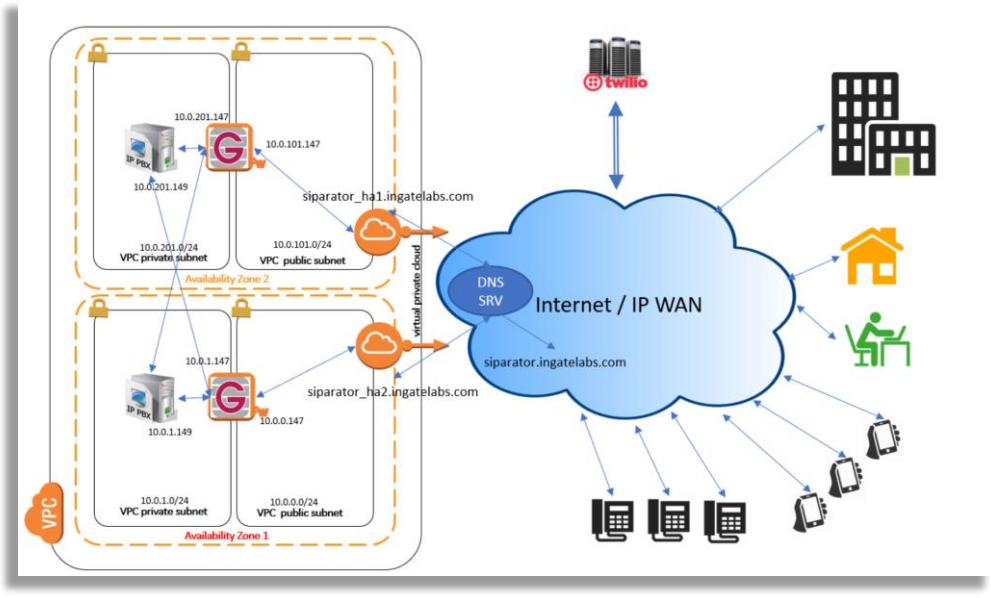


5.4 SIParator configuration for Failover using DNS SRV and other features (Remote Users/Endpoints use case)

In this section, we explain another use case, Remote Users or Endpoints.

The SIParator is the immediate point of contact for any remote SIP device addressable through the Internet. It acts as registrar and proxy, but, it forwards all SIP requests using its built in DNS Override feature.

5.4.1 Using DNS SRV to become the SIP Proxy and Registrar with a Failover with 2 SIParators

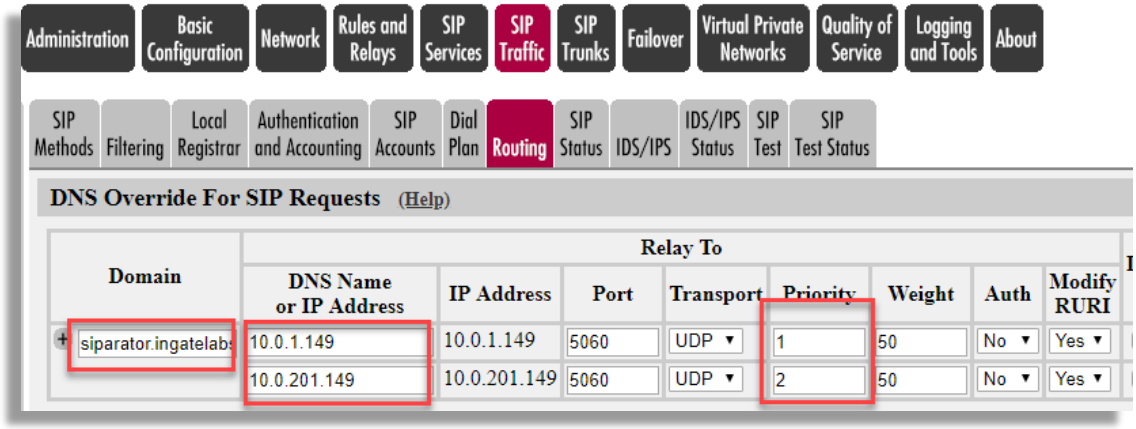


We can create a new SRV record for a different domain name and use the SRV we created in 5.1 as siparator.ingatelabs.com, which we know resolves using Failover or Load Balance Strategy to siparator_ha1.ingatelabs.com and siparator_ha2.ingatelabs.com.

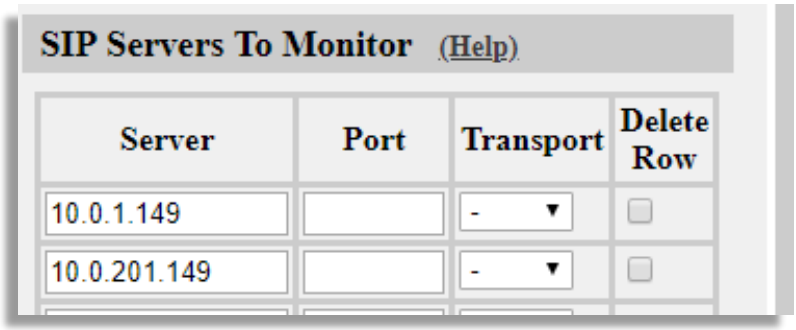
In this way, any SIP request from remote points will be sent to any of the SIParators based on the selected strategy.

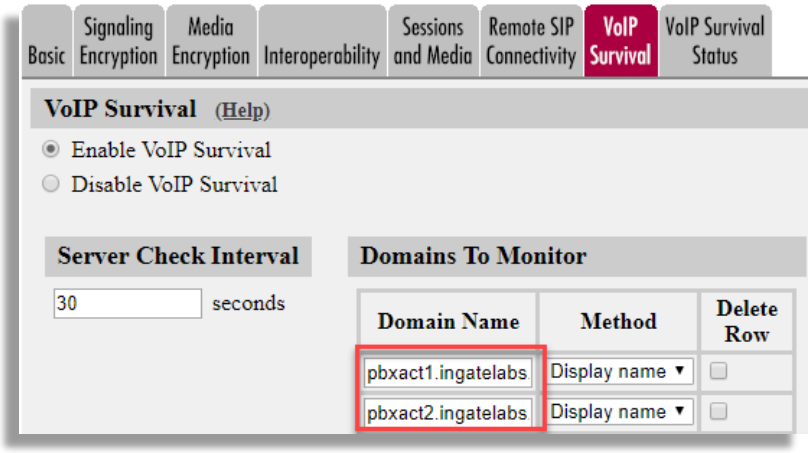
Once the request reaches one of the SIParators, it will use DNS Override to decide which PBX will process the request.

Here is how it looks when using the SIParator to manage PBX Failover strategy. Note we are assigning lower priority to 10.0.1.149, which means that this will be the primary PBX, and only when it's unavailable will the request be sent to the second PBX.



As recommended before we can speed up the detection by adding PBX addresses to the SIP Monitor Server list and including them in the VoIP Survival Table.





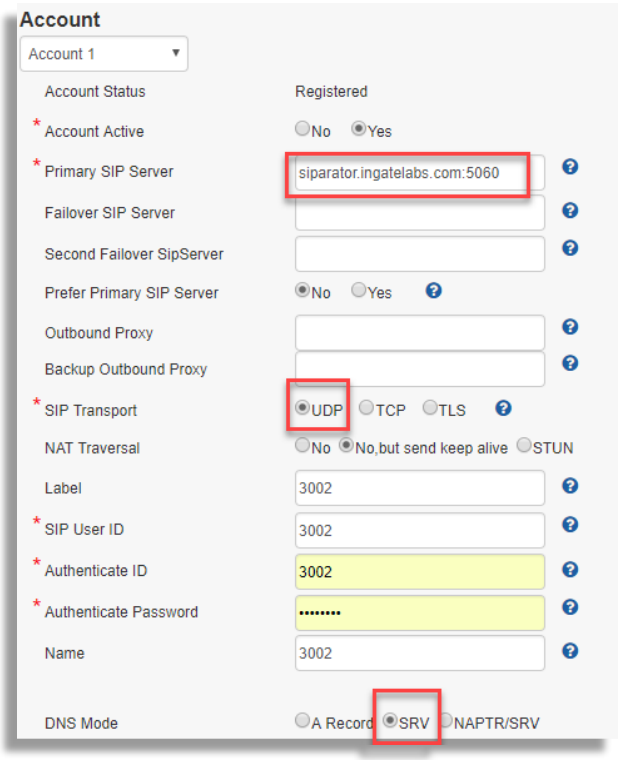
As we have also DNS A Records for a hosted private zone, we can use pbxact1.ingatelabs.local and pbxact2.ingatelabs.local as defined in section 5.2, either on SIP DNS Override, SIP Servers Monitor and VoIP Survival Tables

Your remote users/endpoints will refer then to siparator.ingatelabs.com as Proxy and registrar. You need to make sure the endpoint device supports DNS SRV resolution.

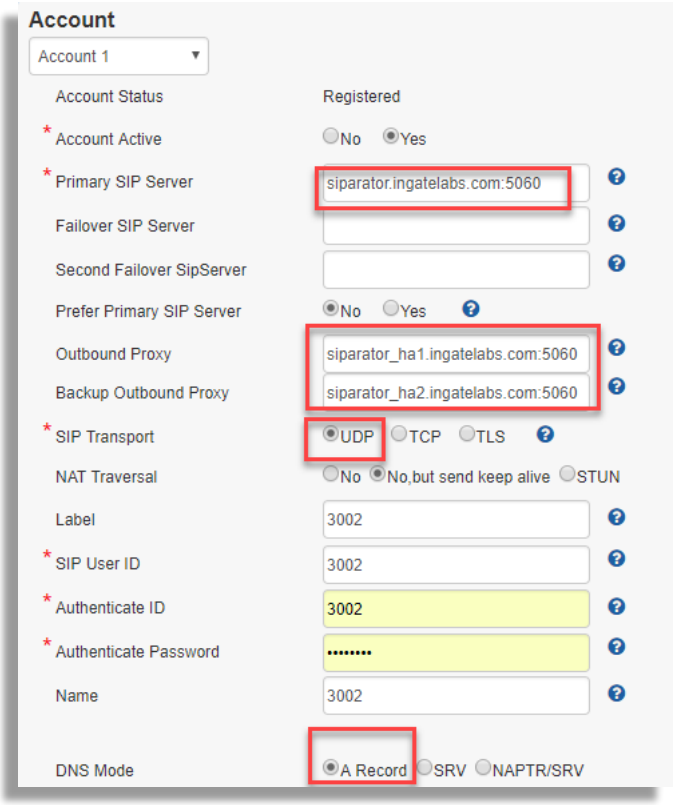
In case the endpoint doesn't support DNS SRV resolution but supports secondary Proxy/Registrar you can use siparator_ha1.ingatelabs.com as the primary and siparator_ha2.ingatelabs.com as the secondary or backup.

Here a few examples:

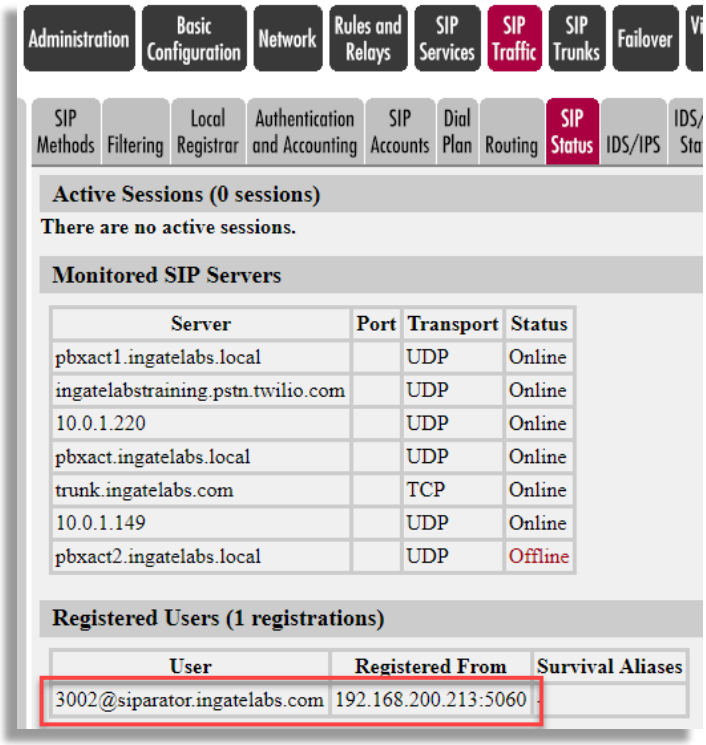
1) Using DNS SRV:



2) Using DNS, A Records (Note the destination for SIP requests is managed by the outbound proxy. The domain is still siparator.ingatelabs.com



Finally, you can visually check registrations in the SIP Status Page on each SIParator:

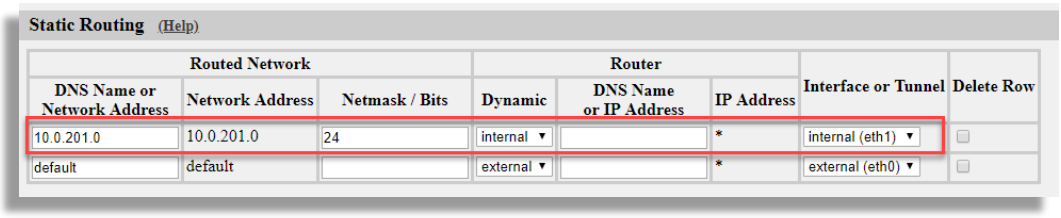


5.5 PBX cross connection with SIParators

One potential scenario when a SIParator only goes offline, or a PBX only goes off line is that sip trunk traffic will go from a Main SIParator to a Failover PBX. In other words will go across in our case subnet 10.0.1.0/24 ←→10.0.201.0/24. This case needs to have extra considerations as none of the SIParators are directly connected to both subnets and we want the traffic between PBX and SIParators always to happen on the Internal Interface.

In this scenario we need to consider adding static routes to each SIParator to make sure the traffic goes from the SIParator internal interface to the appropriate PBX subnet.

On the Main SIParator we will need then to add a static routo to define how to reach the PBX that is on the Failover subnet and not directly connected to it.



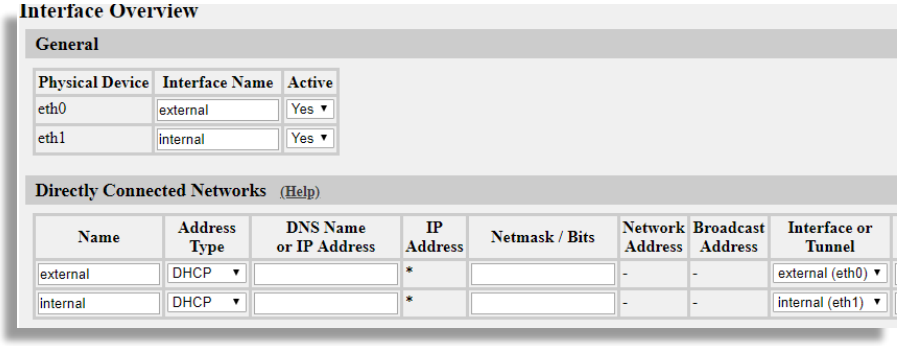
We are saying in order to reach 10.0.201.0 (Private subnet in the failover area), to use the dynamic gateway provided by DHCP on the Internal Interface and route using eth1.

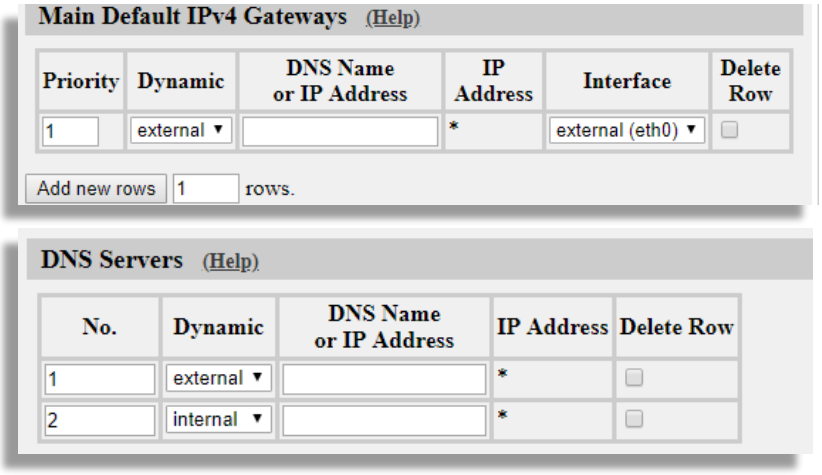
6 Replicating and synchronizing SIParator configurations across all instances

Once you have one of the 2 SIParators fully configured we will need to replicate and synchronize with the secondary/failover configuration.

Some points to clarify:

1) Network Interface configuration is provided via DHCP by AWS VPC





Main Default IPv4 Gateways (Help)

Priority	Dynamic	DNS Name or IP Address	IP Address	Interface	Delete Row
1	external		*	external (eth0)	

Add new rows 1 rows.

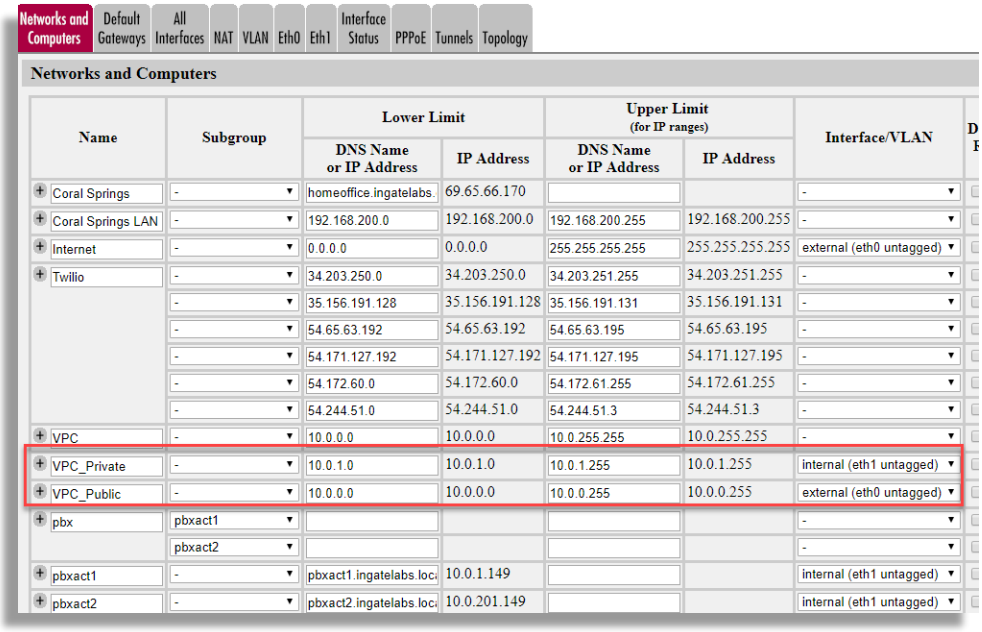
DNS Servers (Help)

No.	Dynamic	DNS Name or IP Address	IP Address	Delete Row
1	external		*	
2	internal		*	

2) Networks and Computers will be very similar in both SIParators. Only VPC_Private and VPC_Public differ from one SIParator to the other

   a. 10.0.0.0/24 → 10.0.10.0/24

   b. 10.0.1.0/24 → 10.0.201.0/24



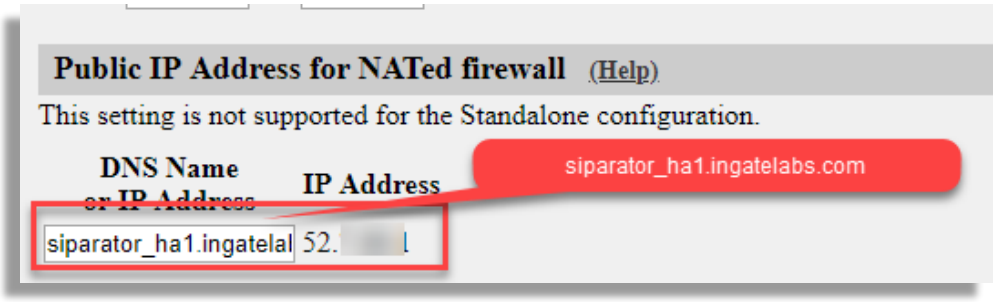
Networks and Computers | Default Gateways | All Interfaces | NAT | VLAN | Eth0 | Eth1 | Interface Status | PPPoE | Tunnels | Topology

Networks and Computers

Name	Subgroup	Lower Limit DNS Name or IP Address	Lower Limit IP Address	Upper Limit (for IP ranges) DNS Name or IP Address	Upper Limit (for IP ranges) IP Address	Interface/VLAN
Coral Springs	-	homeoffice.ingatelabs.	69.65.66.170			-
Coral Springs LAN	-	192.168.200.0	192.168.200.0	192.168.200.255	192.168.200.255	-
Internet	-	0.0.0.0	0.0.0.0	255.255.255.255	255.255.255.255	external (eth0 untagged)
Twilio	-	34.203.250.0	34.203.250.0	34.203.251.255	34.203.251.255	-
	-	35.156.191.128	35.156.191.128	35.156.191.131	35.156.191.131	-
	-	54.65.63.192	54.65.63.192	54.65.63.195	54.65.63.195	-
	-	54.171.127.192	54.171.127.192	54.171.127.195	54.171.127.195	-
	-	54.172.60.0	54.172.60.0	54.172.61.255	54.172.61.255	-
	-	54.244.51.0	54.244.51.0	54.244.51.3	54.244.51.3	-
VPC	-	10.0.0.0	10.0.0.0	10.0.255.255	10.0.255.255	-
VPC_Private	-	10.0.1.0	10.0.1.0	10.0.1.255	10.0.1.255	internal (eth1 untagged)
VPC_Public	-	10.0.0.0	10.0.0.0	10.0.0.255	10.0.0.255	external (eth0 untagged)
pbx	pbxact1					-
	pbxact2					-
pbxact1	-	pbxact1.ingatelabs.loc	10.0.1.149			internal (eth1 untagged)
pbxact2	-	pbxact2.ingatelabs.loc	10.0.201.149			internal (eth1 untagged)

3) To manage Near End Nat Traversal, and because AWS VPC manages Elastic Public IPs as a NAT IP to an interface, we will need to properly assign the Public IP address in SIP Services → Basics. We can use the Domain name for each Specific SIParator as shown here for the Main SIParator in our pair:

SIPARATOR 1



SIPARATOR 2

6.1 How to synchronize Configuration.

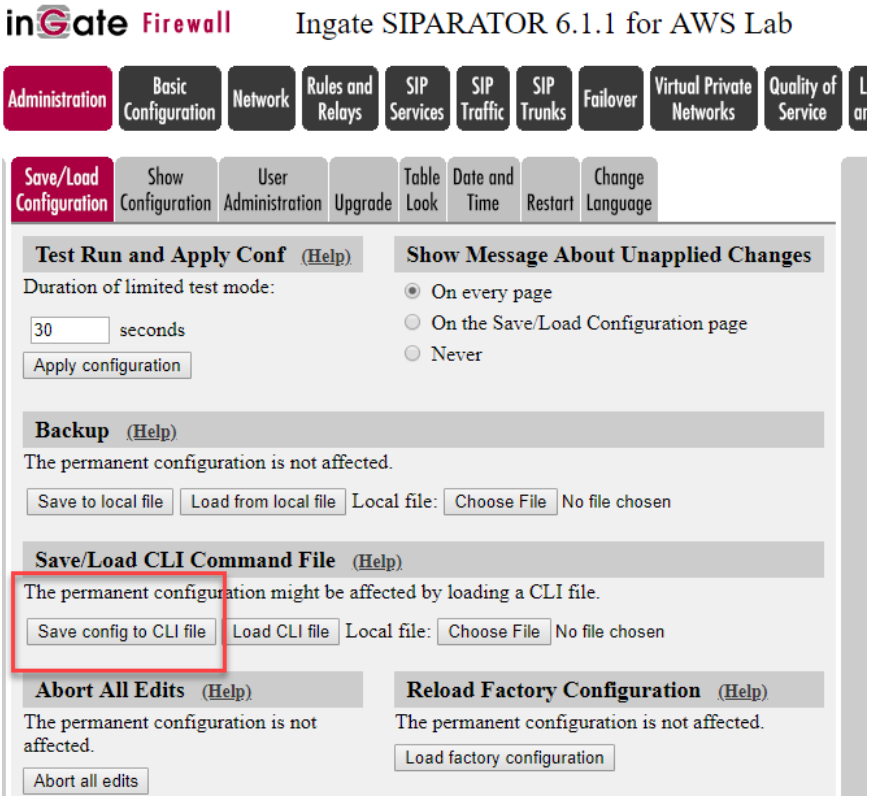
There are two ways we can create and apply the configuration needed and adjusted for the Failover SIParator. Both use the CLI saved file form the Main SIParator.

We can edit even manually (using and editor) or automate using some scripting application the needed changes to be used in the Failover SIParator, or we can just directly load a CLI file in the Failover for the one downloaded from the main unit and do the modifications using the GUI in the second unit.

6.1.1 Obtaining CLI file from the main SIParator

We will manually save a CLI configuration file and use it to load the configuration into the failover unit.

1) Saving the CLI Backup



The file name will be like:

"Ingate SIPARATOR 6.1.1 for AWS Lab_2018-04-27T003837.cli"

6.1.2 Manually modify CLI File:

Using a text editor such as notepad++, make the following changes:

- Modify the Public IP Address for the NATed firewall with the FQDN and IP of the second SIParator:

  Seach for **"# sip.public_ip"**

  You will find something like this:

```
1822
1823 # sip.public_ip
1824 modify-row sip.public_ip 1 ip="siparator_ha1.ingatelabs.com|52.7.99.1|"
1825
```

  Replace this with the FQDN of the second SIParator. No need to change the IP as it will be resolved on the second SIParator once the new configuration is loaded and saved. It should look like this:

```
1822
1823 # sip.public_ip
1824 modify-row sip.public_ip 1 ip="siparator_ha2.ingatelabs.com 52.7.99.1|"
1825
```

- Now we are going to change Private and Public VPC subnets

  Search for **"# firewall.network_groups"**

  You will find something like this:

```
650 # firewall.network_groups
651 clear-table firewall.network_groups
652 add-row firewall.network_groups {id 1} interface=- \
653     lower_ip="homeoffice.ingatelabs.com|69.65.66.170|" \
654     name="Coral Springs" subgroup=- upper_ip=""
655 add-row firewall.network_groups {id 2} interface=- lower_ip=192.168.200.0 \
656     name="Coral Springs LAN" subgroup=- upper_ip=192.168.200.255
657 add-row firewall.network_groups {id 3} interface=eth1 lower_ip=10.0.1.0 \
658     name=VPC_Private subgroup=- upper_ip=10.0.1.255
659 add-row firewall.network_groups {id 4} interface=eth0 lower_ip=10.0.0.0 \
660     name=VPC_Public subgroup=- upper_ip=10.0.0.255
661 add-row firewall.network_groups {id 5} interface=eth0 lower_ip=0.0.0.0 \
662     name=Internet subgroup=- upper_ip=255.255.255.255
663 add-row firewall.network_groups {id 6} interface=- lower_ip=10.0.0.0 \
664     name=VPC subgroup=- upper_ip=10.0.255.255
665 add-row firewall.network_groups {id 8} interface=- lower_ip=54.172.60.0 \
666     name=Twilio subgroup=- upper_ip=54.172.61.255
667 add-row firewall.network_groups {id 9} interface=- lower_ip=54.244.51.0 \
668     name=Twilio subgroup=- upper_ip=54.244.51.3
669 add-row firewall.network_groups {id 10} interface=- lower_ip=54.171.127.192 \
670     name=Twilio subgroup=- upper_ip=54.171.127.195
671 add-row firewall.network_groups {id 12} interface=- lower_ip=35.156.191.128 \
672     name=Twilio subgroup=- upper_ip=35.156.191.131
673 add-row firewall.network_groups {id 13} interface=- lower_ip=54.65.63.192 \
674     name=Twilio subgroup=- upper_ip=54.65.63.195
675 add-row firewall.network_groups {id 15} interface=- lower_ip=34.203.250.0 \
676     name=Twilio subgroup=- upper_ip=34.203.251.255
677 add-row firewall.network_groups {id 16} interface=eth1 \
678     lower_ip="pbxact1.ingatelabs.local|10.0.1.149|" name=pbxact1 \
679     subgroup=- upper_ip=""
```

  Change the Upper and Lower IPs to match the new subnet in the second SIParator. It should then look like this:

```
657 add-row firewall.network_groups {id 3} interface=eth1 lower_ip=10.0.201.0 \
658     name=VPC_Private subgroup=- upper_ip=10.0.201.255
659 add-row firewall.network_groups {id 4} interface=eth0 lower_ip=10.0.101.0 \
660     name=VPC_Public subgroup=- upper_ip=10.0.101.255
```

- As we are using Twilio, we will change the Host name in Request URI to the Public IP address of the Secondary SIParator

  Search for **"domain_id"**

```
2135 # sipswitch.trunk_params
2136 modify-row sipswitch.trunk_params 1 alias_ip=- \
2137     domain=ingatelabstraining.pstn.twilio.com domain_id=52      .1 \
2138     enabled=off from_domain=pdomain from_domain_str="" fwd_refer=off \
2139     gin_reg=off hide_rr=off hide_to_tags=off itsp_host_addrs=Twilio \
2140     ltrunk_group_param="" ltrunk_group_usage=- max_calls_per_line="" \
2141     max_calls_total=10 name="Twilio Elastic" outbound_gw=- \
2142     outbound_proxy="" port="" preserve_max_forwards=off \
2143     redirect_caller_domain=off redirect_home_domain=off \
2144     referto_domain="" relay_media=on remove_via=off route_incoming=ruri \
2145     send_dtmf_via_sip_info=off transport=udp trunk=1 \
2146     trunk_group_param="" trunk_group_usage=- trusted_networks_enable=off \
2147     use_preferred_identity=off
```

  Change the value for domain_id to the public IP address of the destination SIParator. It should look like this:

```
2135 # sipswitch.trunk_params
2136 modify-row sipswitch.trunk_params 1 alias_ip=- \
2137     domain=ingatelabstraining.pstn.twilio.com domain_id=35.          .228 \
2138     enabled=off from_domain=pdomain from_domain_str="" fwd_refer=off \
2139     gin_reg=off hide_rr=off hide_to_tags=off itsp_host_addrs=Twilio \
2140     ltrunk_group_param="" ltrunk_group_usage=- max_calls_per_line="" \
2141     max_calls_total=10 name="Twilio Elastic" outbound_gw=- \
2142     outbound_proxy="" port="" preserve_max_forwards=off \
2143     redirect_caller_domain=off redirect_home_domain=off \
2144     referto_domain="" relay_media=on remove_via=off route_incoming=ruri \
2145     send_dtmf_via_sip_info=off transport=udp trunk=1 \
2146     trunk_group_param="" trunk_group_usage=- trusted_networks_enable=off \
2147     use_preferred_identity=off
```

- Need to adjust Dial Plan for Matching Request-URI

  In case you are matching the internal IP address in the Request URI in the dial plan you might need to consider this change in the file to be synchronized.

  Seach for **"# sipswitch.request_to"**

```
2101 # sipswitch.request_to
2102 clear-table sipswitch.request_to
2103 add-row sipswitch.request_to {id 1} domain=10.0.1.147 head="" \
2104     min_tail_length="" name=to_pstn prefix="" regexp="" tail=telchar
2105
```

  Replace the IP with the Failover internal IP address:

```
2101 # sipswitch.request_to
2102 clear-table sipswitch.request_to
2103 add-row sipswitch.request_to {id 1} domain=10.0.201.147 head="" \
2104     min_tail_length="" name=to_pstn prefix="" regexp="" tail=telchar
2105
```

  In order to avoid this change you can build the Rules in the dial plan using regular expressions to match ip addresses of all SIParators used in failover. You can use something similar to this rule in the Match Request-URI:

Matching Request-URI (Help)

Name	Use This ...					... Or This	Delete Row
	Prefix	Head	Tail	Min. Tail	Domain	Reg Expr	
to_pstn			-			sip:(.*)@10\.0\.(1|201)\.147	

- Replicate static route to consider Failover and non-failover cross connection

  Search for **"# network.routes"**

```
1342 # network.routes
1343 clear-table network.routes
1344 add-row network.routes {id 1} destination=default/ gateway="|*||external" \
1345     interface=eth0 priority=1
1346 add-row network.routes {id 2} destination=10.0.201.0/24 \
1347     gateway="|*||internal" interface=eth1 priority=""
1348
```

Notice The Primary SIParator has a static route to the subnet where the failover PBX is located.

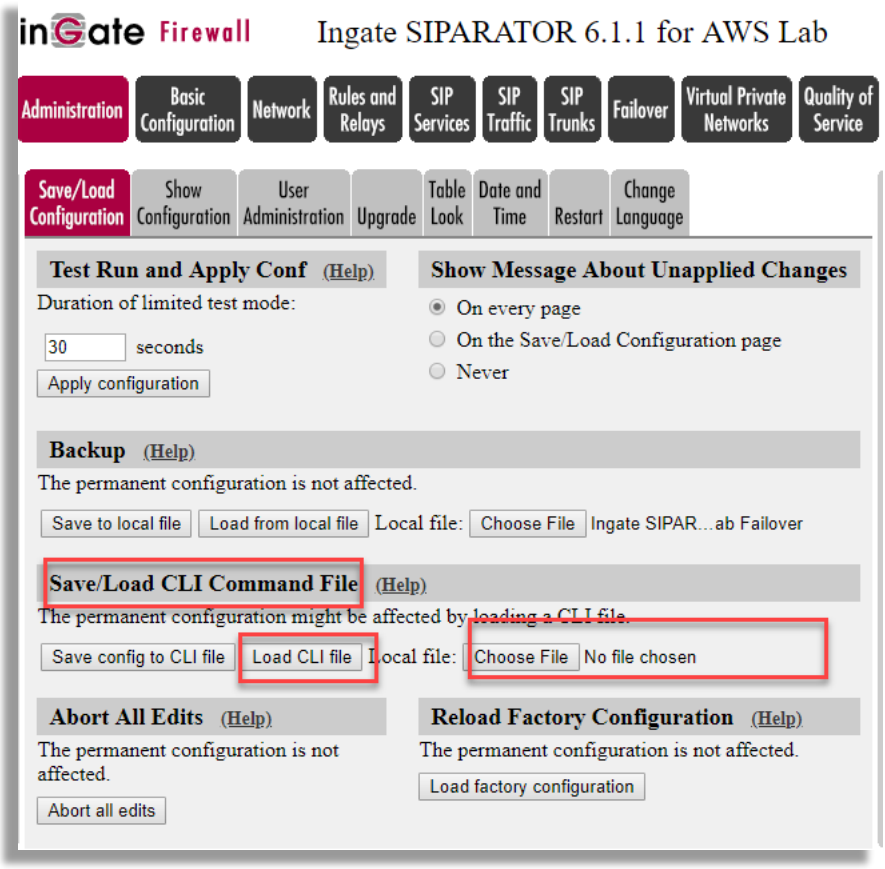
We will need to put on the failover SIParator a static route to the subnet where the main PBX is located. Like this:

```
1342 # network.routes
1343 clear-table network.routes
1344 add-row network.routes {id 1} destination=default/ gateway="|*||external" \
1345     interface=eth0 priority=1
1346 add-row network.routes {id 2} destination=10.0.1.0/24 \
1347     gateway="|*||internal" interface=eth1 priority=""
1348
```

- Save your changes to the file

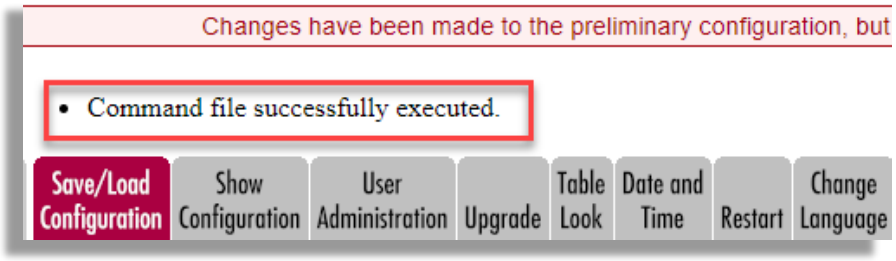
2) Load the Modified CLI File.

Once you have logged in to the Failover SIParator go to the Administration → Save/Load Configuration and load the CLI file you modified in the previous step.



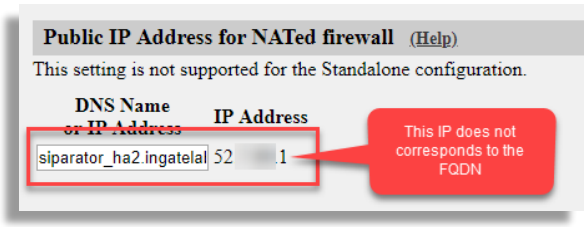
- Chose the file
- Load CLI File

Once the file is loaded you'll get a confirmation message like this:

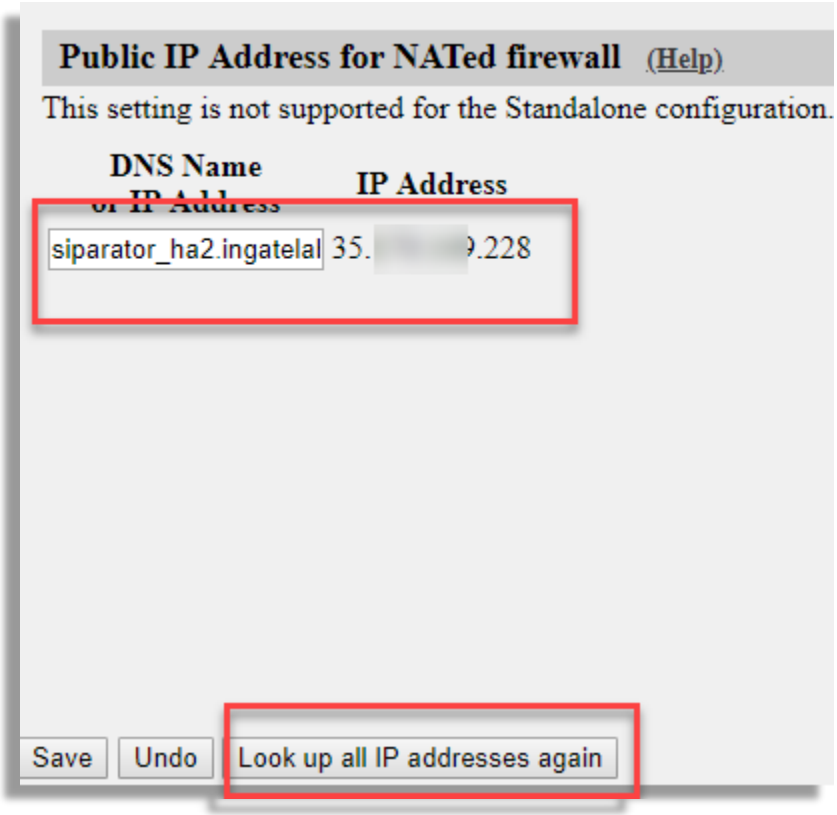


Now, go to SIP Services → Basic and let's refresh the Public IP address associated to the FQDN we entered in the CLI file.

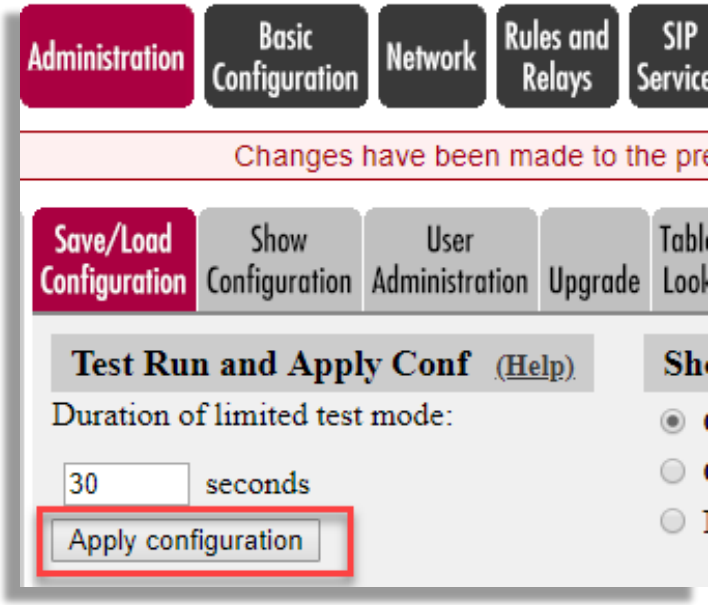
It looks like:

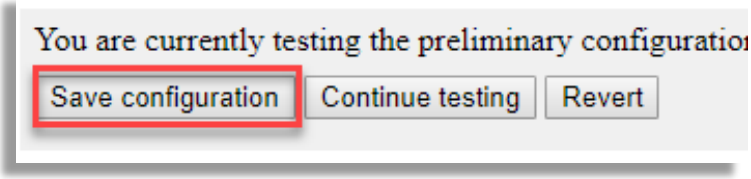


Using the "Look up All IP addresses again" button it will be resolved again and updated correctly.



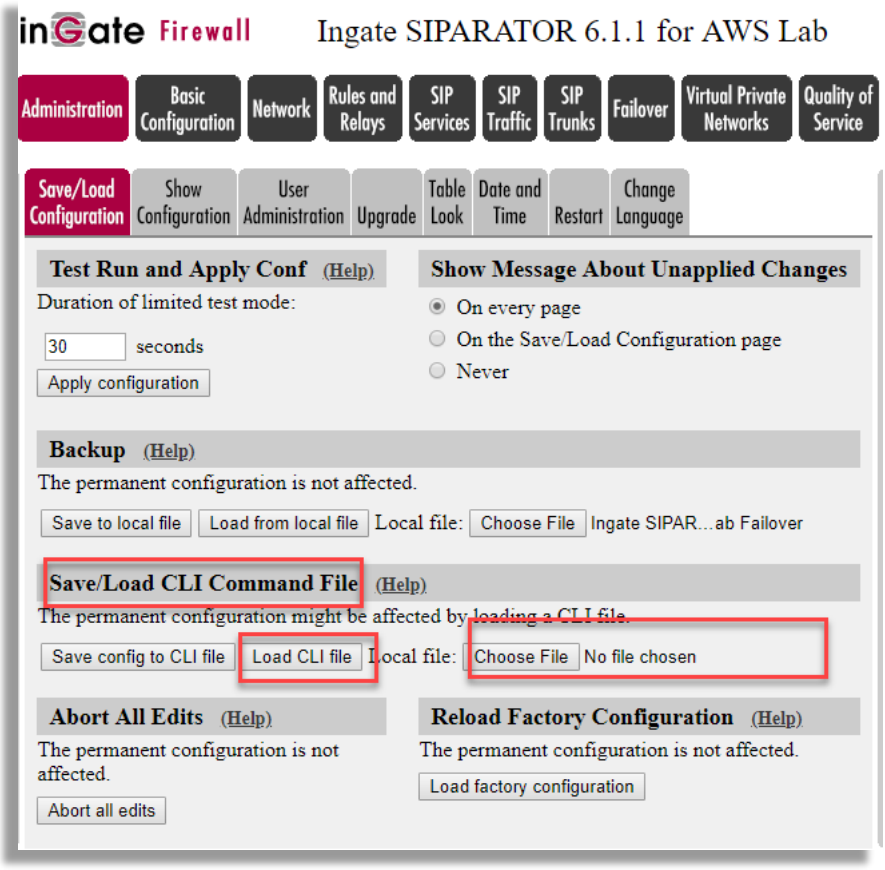
Next step is just to Apply all changes and save.





6.1.3 Loading CLI file and use the GUI to make all needed adjustments

Once you have logged in to the Failover SIParator go to the Administration → Save/Load Configuration and load the CLI file you downloaded from the Main SIParator.



- Chose the file
- Load CLI File

Once the file is loaded you'll get a confirmation message like this:

At this point you have exactly the same configuration you have on the Main unit, loaded in the Failover unit, but it hasn't been applied yet.

Before applying the configuration, we will modify all the parameter needed.

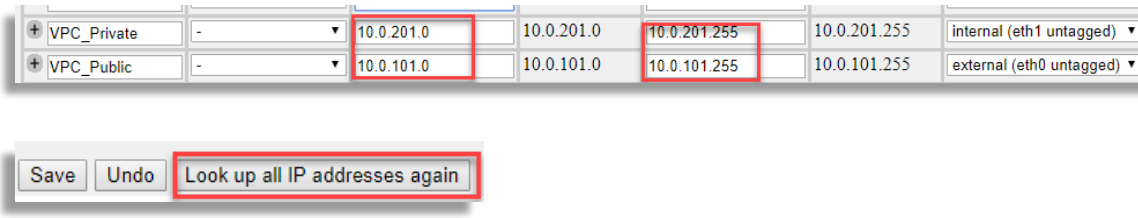
- Modify the Public IP Address for the NATed firewall with the FQDN and IP of the second SIParator:

  SIP Services → Basic

  Change the FQDN to the one associated to the Public IP address of the Failover SIParator. Then press "Look up IP addresses again" to discover the new IP address

- Now we are going to change Private and Public VPC subnets

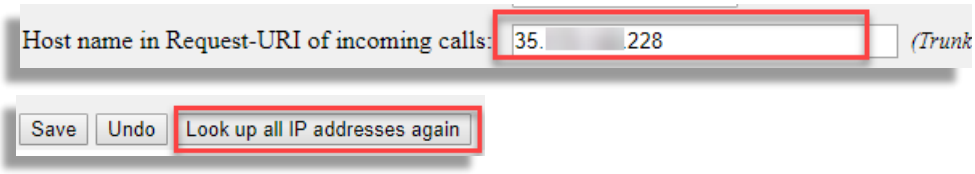
  Networks → Networks and Computers



  Change the Upper and Lower IPs to match the new subnet in the second SIParator. Then press "Look up IP addresses again" to discover the new IP address

- As we are using Twilio, we will change the Host name in Request URI to the Public IP address of the Secondary SIParator

  SIP Trunks → Go to SIP Trunk



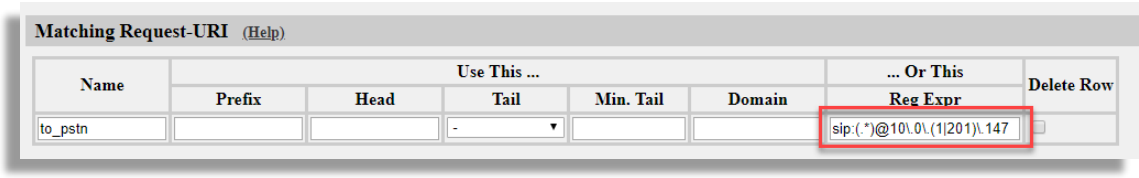
- Need to adjust Dial Plan for Matching Request-URI

  In case you are matching the internal IP address in the Matching Request URI in the dial plan you might need to consider this change in the file to be synchronized.

  SIP Traffic →Dial Plan

  Replace the IP with the Failover internal IP address:

  Here we can build the Rules in the dial plan using regular expressions to match ip addresses of all SIParators used in failover with a single rule. You can use something similar to this rule in the Match Request-URI:



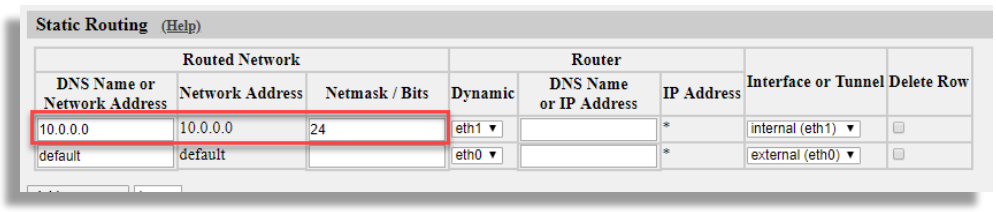
- Replicate static route to consider Failover and non-failover cross connection

  Network → All Interfaces

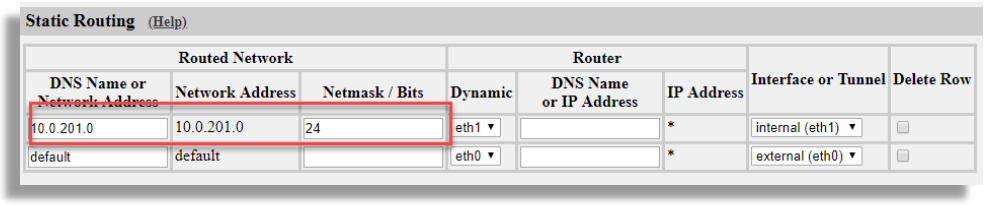
  Notice The Primary SIParator has a static route to the subnet where the failover PBX is located.

  We will need to put on the failover SIParator a static route to the subnet where the main PBX is located. Like this:

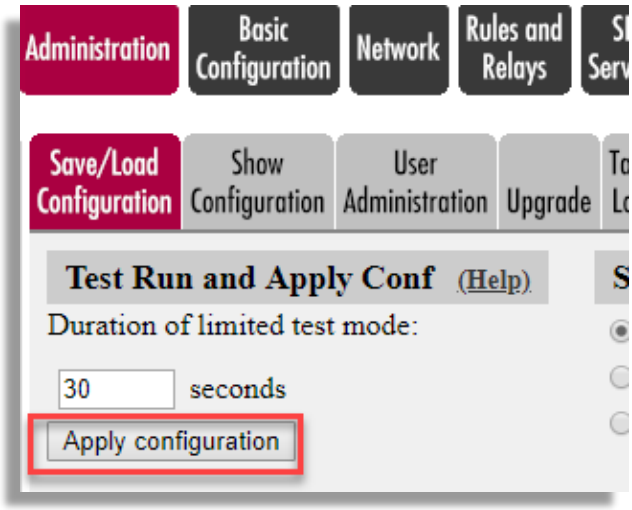
  Failover Static Routes:

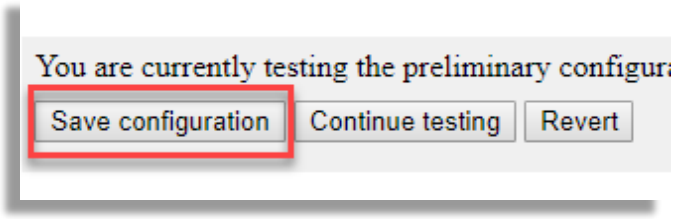


Main Static Route



- Save all and apply changes





7 Final Tips and recommendations

There is not a "one size fits all" solution when talking about High Availability (HA) deployments in AWS, but the structure presented here is a good template to start and build variations on top of it.

What is presented here considers 2 SIParators in Failover or Load Balance, inside the same VPC but in separated Availability zones. This is a typical case when we want to prevent not only Instance failures but Data Center Isolations.

With minor adjustments it can be extrapolated to other scenarios, such as:

- Two SIParators in separated VPCs. In this case, some additional modifications will be needed in the CLI Migration file, specifically in the Networks and Computers section. Also, you will need to enable peer connection between VPCs using native services from AWS, and some additional Route53 Hosted Private Zone and SRV records.
- Two SIParators in separated regions. This case is very similar to Separated VPCs in the same region

One recommended reading about VPC Peering can be found here:

https://docs.aws.amazon.com/AmazonVPC/latest/PeeringGuide/Welcome.html

The following table can help identify how HA can be implemented depending on the scenarios and what we want, in order to prevent specific call flow interruptions. (see next page)

TOOLS TO BE USED	TECHNIQUE	SIP TRUNKING FLOWS				REMOTE USERS FLOWS			
		PBX --> SBC Outbound	PBX <--SBC Inbound	SBC-->PSTN Outbound	SBC<--PSTN Inbound	PBX --> SBC Outbound	PBX <--SBC Inbound	SBC-->INET Outbound	SBC<--INET Inbound
	SIP Filtering	Need to add IPPBX IP addresses or DNS names for each PBX in the Networks and computers section and enable processing calls from that name in SIP Traffic Filters	NA	NA	Need to add ITSP potential souce IP addresses including SIP and RTP in Network and computers section and enable processings calls from that name in SIP Traffic Filters	NA	NA	NA	Need to add all portential Source IP address form which calls can be originated in the Networks and Computers section and enable processing calls for those names in the SIP Traffic Filters. This might represent a security breach if source IP's are unpredictable. In this case, it Is recomended to use TLS for
	PBX Dual Homing	NA	Use SIParator native capability in SIP Trunk Group to associate up to 2 domains to a PBX (see	NA	NA	NA	NA	NA	NA
	SIP Trunk Dual Homming	NA	NA	Use SIParator native capability in SIP Trunk Group to associate up to 2 domains to an ITSP	NA	NA	NA	NA	NA
	DNS SRV Public Hosted Zones	NA	NA		If ITSP Supports DNS SRV, would be enough to provide Failover or load balance destinations to the SBC from the Carrier.	NA	NA	NA	Any Remote endpoint will use (Registrar & Proxy) domain name defined in SRV Record. DNS will resolve failover or balancing depending on the
	DNS SRV Private Hosted Zones	If PBX supports resolving DNS SRV, a single trunk will be enough to reach any of the SBC, either for Failover or load balance strategy	Defining SRV Records to reach the PBX will allow to establish a Failover or Load Balance strategy to reach any PBX associated to the same internal/private	NA	NA	Defining SRV Records to reach Internaly the SBC will allow to establish a Failover or Load Balance strategy to reach any PBX associated to the same	Defining SRV Records to reach Internallythe PBX will allow to establish a Failover or Load Balance strategy to reach any PBX associated to the same	NA	NA
	DNS Override (SIParator)	NA	NA	NA	NA	NA	Here DNS Override will act as an internal DNS and will be able to route SIP Requests coming from outside to the corresponding	As part of DNS Override a local registartio table is kept for location purposes of the end points	NA
	Dial Plan (SIParator)	NA	NA	Dial Plan is an additional resource to add complementary techniques to associate more than one destination and hunt between them for a given call	NA	NA	NA	NA	NA
	VoIP Survival (SIParator)	NA	NA	NA	NA	NA	Using VoIP Survival, in case no PBX is available, SiParator can asume a temporary rol to become the Registrar and Sproxy	NA	Using VoIP Survival, in case no PBX is available, SiParator can asume a temporary rol to become the Registrar and Sproxy

8 Additional help or support

If you have questions, suggestions and any other concern feel free to contact Educronix LLC

Web: www.educronix.com
Email: support@educronix.com
Toll-Free: +1 855 866 8854
Ph: +1 954 866 8884

We also provide consulting services as well as remote hands troubleshooting and configuration.