



Application Notes for Mutare Voice Spam Filter with Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Mutare Voice Spam Filter to interoperate with Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise. Mutare Voice Spam Filter is a call filtering solution.

In the compliance testing, Mutare Voice Spam Filter used SIP trunk with Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise to support spam call filtering.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for Mutare Voice Spam Filter to interoperate with Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise (SBCE). Voice Spam Filter is a call filtering solution.

In the compliance testing, Voice Spam Filter used SIP trunk with Session Manager and SBCE to support spam call filtering.

Voice Spam Filter can be deployed as a standalone solution or as a feature of the Mutare Voice solution. The compliance testing focused on Voice Spam Filter as a standalone call filtering solution.

Incoming calls to the Avaya SIP-enabled network are delivered by SBCE via SIP trunk to Voice Spam Filter for spam call filtering. Voice Spam Filter examines the SIP call signaling information to identify the caller ID, and checks the caller ID against enterprise whitelist, enterprise blacklist, as well as dynamic robocall list hosted on the Mutare external database in the cloud. Non-spam calls are released by Voice Spam Filter to Session Manager, and spam calls can be configured to be dropped or redirected to resource destinations on Communication Manager. Released and redirected calls are accomplished by modifying the SIP INVITE request line and sent to Session Manager as the next hop.

The Voice Spam Filter solution consisted of a Voice Screening Proxy server and a Voice Application Server. The Voice Screening Proxy was the server that interfaced with Session Manager and SBCE via SIP trunk. The Voice Application Server checked the caller ID against the local enterprise whitelist and blacklist and interfaced with the Mutare cloud for check of caller ID against the dynamic robocall list on the external database.

2. General Test Approach and Test Results

The feature test cases were performed manually. Inbound calls were made from different PSTN calling numbers that match to the enterprise whitelist, enterprise blacklist, dynamic robocall list on external database, along with different settings for spam call handling.

The serviceability test cases were performed manually such as disconnecting/reconnecting the Ethernet connection to Voice Spam Filter.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Voice Spam Filter did not include use of any specific encryption features as requested by Mutare.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following on Voice Spam Filter:

- Proper handling of SIP exchanges including OPTIONS, G.711MU, G.729, codec negotiation, media shuffling, and session refresh.
- Proper handling of call scenarios including release, redirect, blacklist, whitelist, robocall list, not on any list, hold/resume, forwarding, transfer, conference, abandon, invalid number, do not disturb, busy, and simultaneous calls.

The serviceability testing focused on verifying the ability of Voice Spam Filter to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to Voice Screening Proxy, and of SBCE to activate alternate route to Session Manager when Voice Screening Proxy did not respond within the specified interval.

2.2. Test Results

All test cases were executed, and the following were observations on Voice Spam Filter:

- By design, only SIP signaling packets flow through Voice Spam Filter and not RTP packets.
- By design, the first call for the day or the call after Voice Application Server has been idling for a while can take longer for Voice Spam Filter to process. In the compliance testing, the experienced delay was ~7 seconds from the time Voice Spam Filter received the INVITE to the time the message was released to Session Manager.
- An updated opensips.cfg script dated 8/22/2019 is needed to replace the default version that came with Voice Screening Proxy version 2.4.5. The updated script included fixes for redirected calls and for Voice Screening Proxy to stay in the record route until end of call.
- For a call scenario where the SIP Service Provider sent a session interval deemed insufficient by Communication Manager with a 422 Session Interval Too Small being exchanged and therefore a subsequent re-INVITE, Voice Spam Filter reported two history entries for the scenario. This can be managed by ensuring the SIP Service Provider is not sending session intervals that are too small as part of initial planning.

2.3. Support

Technical support on Voice Spam Filter can be obtained through the following:

- **Phone:** +1 (855) 782-3890
- **Email:** help@mutare.com
- **Web :** <http://www.mutare.com/support.asp>

3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**.

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, Session Manager, and SBCE are not the focus of these Application Notes and will not be described.

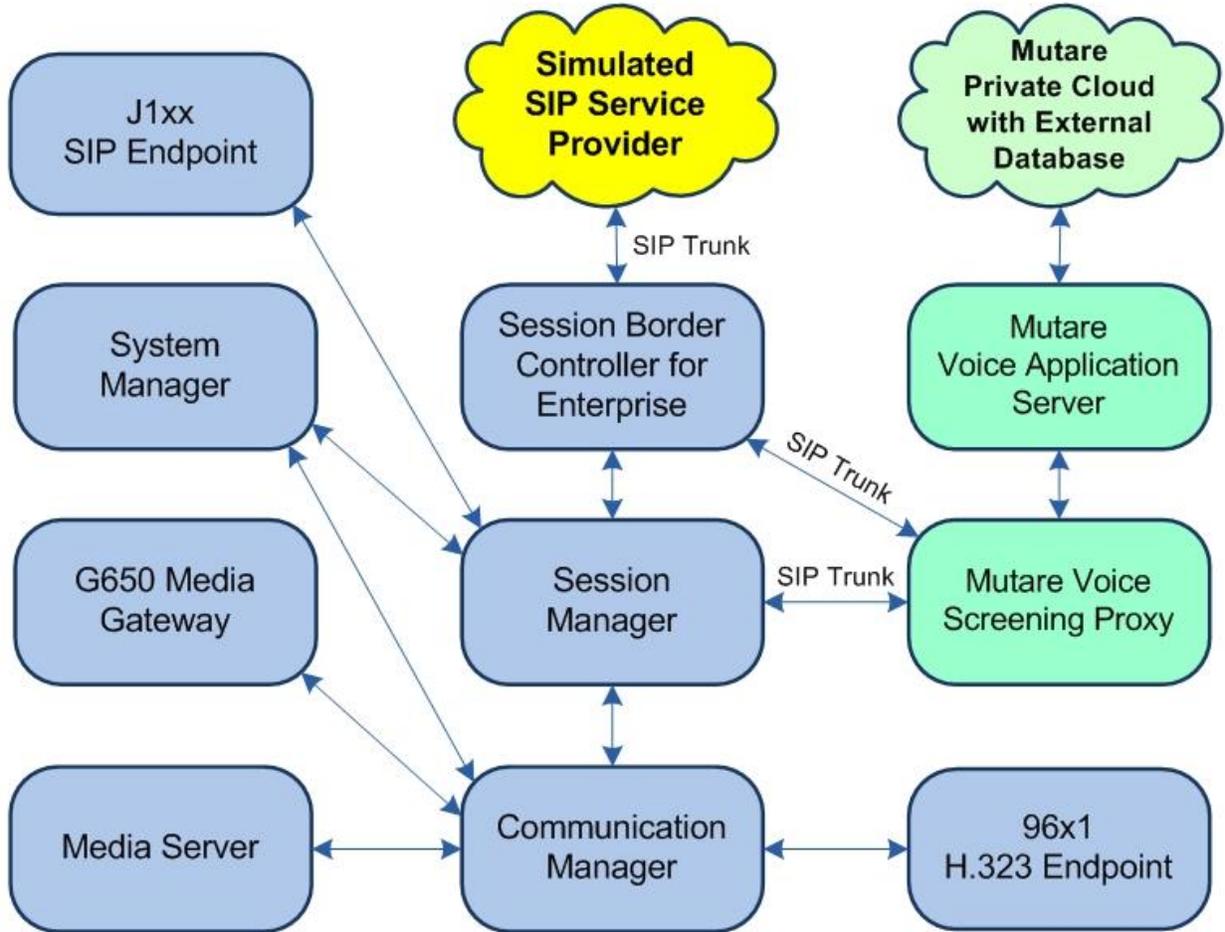


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in Virtual Environment	8.1 (8.1.0.1.1.890.25517)
Avaya G650 Media Gateway	NA
Avaya Aura® Media Server in Virtual Environment	8.0.1.121
Avaya Aura® Session Manager in Virtual Environment	8.1 (8.1.0.0.810007)
Avaya Aura® System Manager in Virtual Environment	8.1 (8.1.0.0.079814)
Avaya Session Border Controller for Enterprise in Virtual Environment	8.0 (8.0.0.0-19-16991)
Avaya 9611G & 9641G IP Deskphone (H.323)	6.8202
Avaya J129 IP Deskphone (SIP)	4.0.2.1.3
Mutare Voice Screening Proxy on CentOS <ul style="list-style-type: none">• opensips.cfg	2.4.5 7 8/22/2019
Mutare Voice Application Server on Windows Server 2016	1.9.0.0 Standard

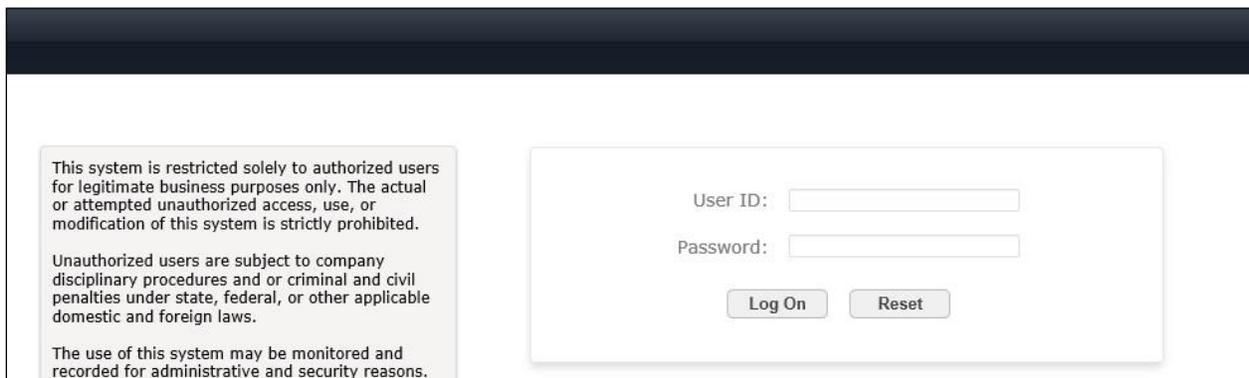
5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer SIP entities

5.1. Launch System Manager

Access the System Manager web interface by using the URL <https://ip-address> in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons.

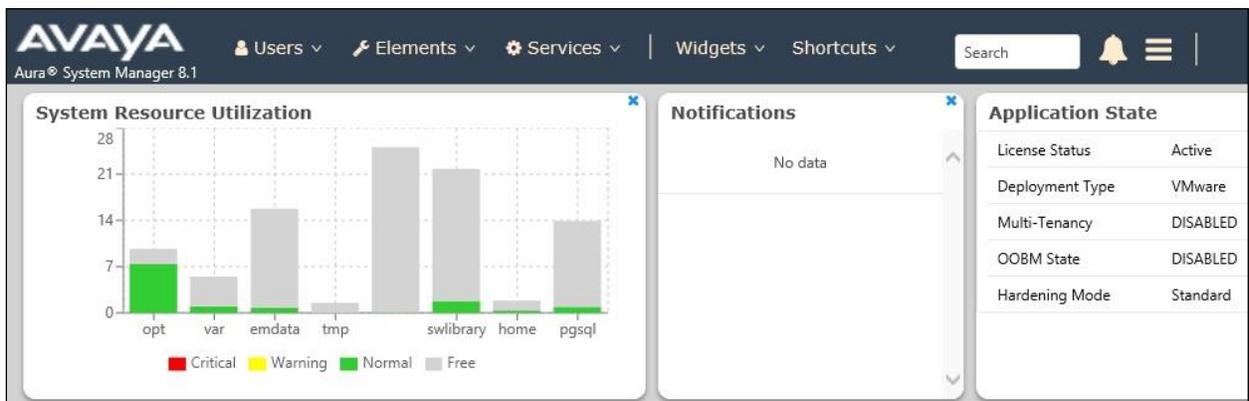
User ID:

Password:

Log On Reset

5.2. Administer SIP Entities

The screen below is displayed.



5.2.1. SIP Entity for Voice Spam Filter

Select **Elements** → **Routing** → **SIP Entities** from the top menu, followed by **New** in the subsequent screen (not shown) to add a new SIP entity for Voice Spam Filter.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the Voice Screening Proxy server.
- **Type:** “SIP Trunk”
- **Notes:** Any desired notes.
- **Location:** Select the pertinent pre-existing location name.
- **Time Zone:** Select the applicable time zone.

The screenshot displays the 'SIP Entity Details' configuration page in Avaya Aura System Manager 8.1. The page is organized into sections: **General**, **Loop Detection**, and **Monitoring**. The **General** section includes fields for Name (Mutare), FQDN or IP Address (10.64.101.203), Type (SIP Trunk), Notes (Voice Screening Proxy), Adaptation, Location (DR-Loc), and Time Zone (America/New_York). The **Loop Detection** section includes Loop Detection Mode (On), Loop Count Threshold (5), and Loop Detection Interval (200). The **Monitoring** section includes SIP Link Monitoring, CRLF Keep Alive Monitoring (both set to Use Session Manager Configuration), and Supports Call Admission Control (unchecked). The interface also features a top navigation bar with 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, a search bar, and a 'Help ?' link. A left-hand navigation menu shows 'SIP Entities' as the active selection. 'Commit' and 'Cancel' buttons are located in the top right corner of the main content area.

Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DR-SM”.
- **Protocol:** “TCP”
- **Port:** “5060”
- **SIP Entity 2:** The Voice Spam Filter entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

Note that Voice Spam Filter can support UDP and TCP, and the compliance testing used the TCP protocol.

Entity Links

Override Port & Transport with DNS SRV:

Add Remove

1 Item Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	*SM-Mutare	DR-SM	TCP	*5060	Mutare	*5060	trusted	<input type="checkbox"/>

Select : All, None

SIP Responses to an OPTIONS Request

Add Remove

0 Items Filter: Enable

Response Code & Reason Phrase	Mark Entity Up/Down	Notes
-------------------------------	---------------------	-------

Commit Cancel

5.2.2. SIP Entity for Session Manager

The **SIP Entities** screen is displayed again. Select the entry associated with Session Manager, in this case “DR-SM”.

<input type="checkbox"/>	Name	FQDN or IP Address	Type	Notes
<input type="checkbox"/>	ACCS1-IP500V2	10.64.125.130	SIP Trunk	
<input type="checkbox"/>	DR-CM	10.64.101.236	CM	TLT DR CM
<input type="checkbox"/>	DR-CM-5212	10.64.101.236	CM	CM Port 5212 (SBCE for SBC-IPOSE)
<input type="checkbox"/>	DR-MSG	10.64.101.224	Messaging	
<input checked="" type="checkbox"/>	DR-SM	10.64.101.238	Session Manager	TLT DR SM
<input type="checkbox"/>	IPO1-IP500V2	192.168.200.134	SIP Trunk	
<input type="checkbox"/>	IPO2-IP500V2	192.168.200.234	SIP Trunk	
<input type="checkbox"/>	IPO2-IPOSE	10.64.101.234	SIP Trunk	
<input type="checkbox"/>	Mutare	10.64.101.203	SIP Trunk	Voice Screening Proxy
<input type="checkbox"/>	SBCE	10.64.101.221	SIP Trunk	

The **SIP Entity Details** screen is displayed next, as shown below.

SIP Entity Details

General

* Name:

* IP Address:

SIP FQDN:

Type:

Notes:

Location:

Scroll down to the **Listen Ports** sub-section and make certain that Session Manager is listening on the transport protocol used by Voice Spam Filter from **Section 5.2.1**, in this case “TCP” as shown below.

Failover Ports

TCP Failover port:

TLS Failover port:

Listen Ports

Add Remove

3 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	TCP ▾	dr220.com ▾	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	UDP ▾	dr220.com ▾	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	TLS ▾	dr220.com ▾	<input checked="" type="checkbox"/>	<input type="text"/>

Select : All, None

SIP Responses to an OPTIONS Request

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
<input type="checkbox"/>			

Commit Cancel

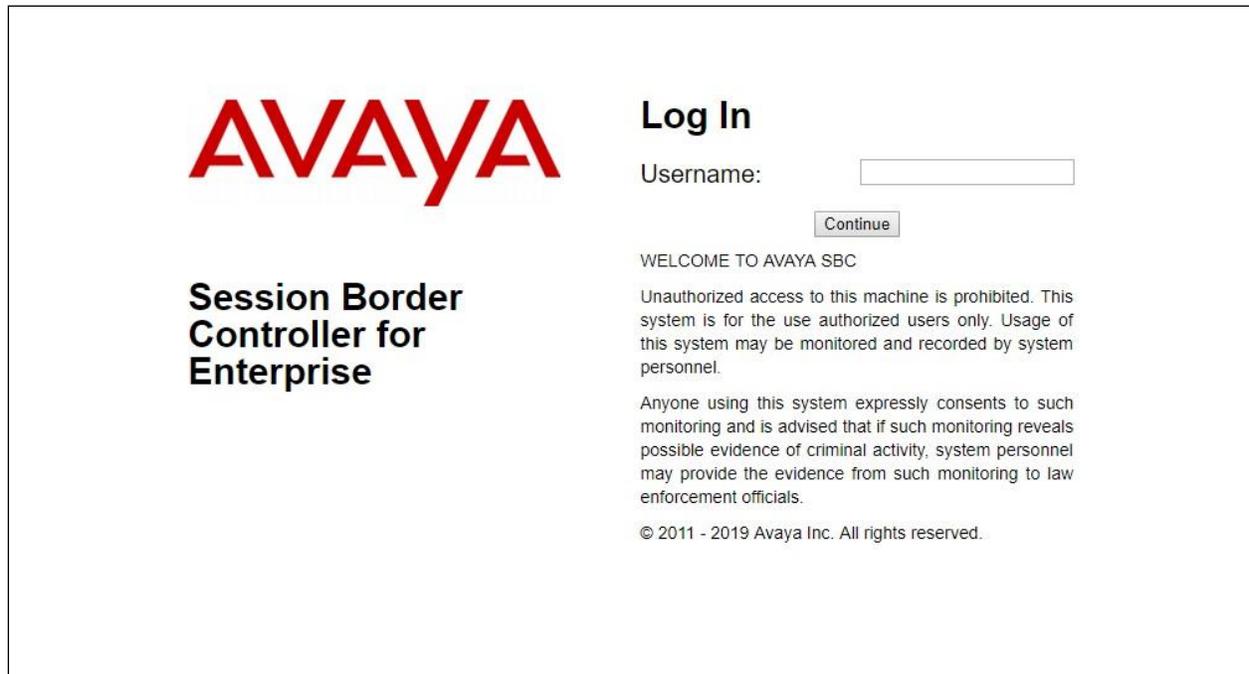
6. Configure Avaya Session Border Controller for Enterprise

This section provides the procedures for configuring SBCE. The procedures include the following areas:

- Launch web interface
- Administer SIP server profile
- Administer routing profile
- Administer interworking profile

6.1. Launch Web Interface

Access the SBCE web interface by using the URL “https://ip-address/sbc” in an Internet browser window, where “ip-address” is the IP address of the SBCE management interface. The screen below is displayed. Log in using the appropriate credentials.



The screenshot shows the Avaya Session Border Controller for Enterprise login page. On the left, the Avaya logo is displayed in red, with the text "Session Border Controller for Enterprise" below it. On the right, the "Log In" section contains a "Username:" label, a text input field, and a "Continue" button. Below the login fields, there is a "WELCOME TO AVAYA SBC" message, a warning about unauthorized access, a consent statement, and a copyright notice: "© 2011 - 2019 Avaya Inc. All rights reserved."

6.2. Administer SIP Server Profile

In the subsequent screen, select **Device** → **SBCE** from the left top menu, followed by **Backup/Restore** → **Services** → **SIP Servers** from the left pane to display the existing SIP server profiles.

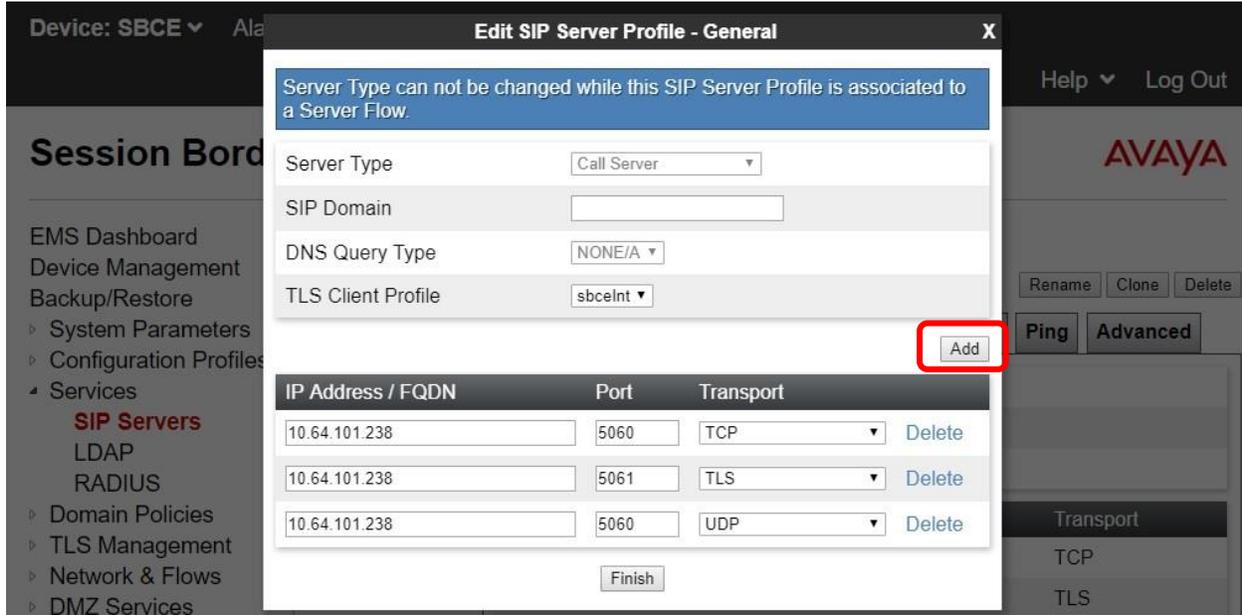
Select the SIP server profile associated with Session Manager, in this case “Server-SM” as shown below. Click **Edit**.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Device: SBCE', 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header displays 'Session Border Controller for Enterprise' and the 'AVAYA' logo. On the left, a navigation menu lists various configuration options, with 'SIP Servers' highlighted under the 'Services' section. The main content area is titled 'SIP Servers: Server-SM' and features an 'Add' button and three action buttons: 'Rename', 'Clone', and 'Delete'. Below this, there are tabs for 'General', 'Authentication', 'Heartbeat', 'Registration', 'Ping', and 'Advanced', with 'General' selected. The configuration details are as follows:

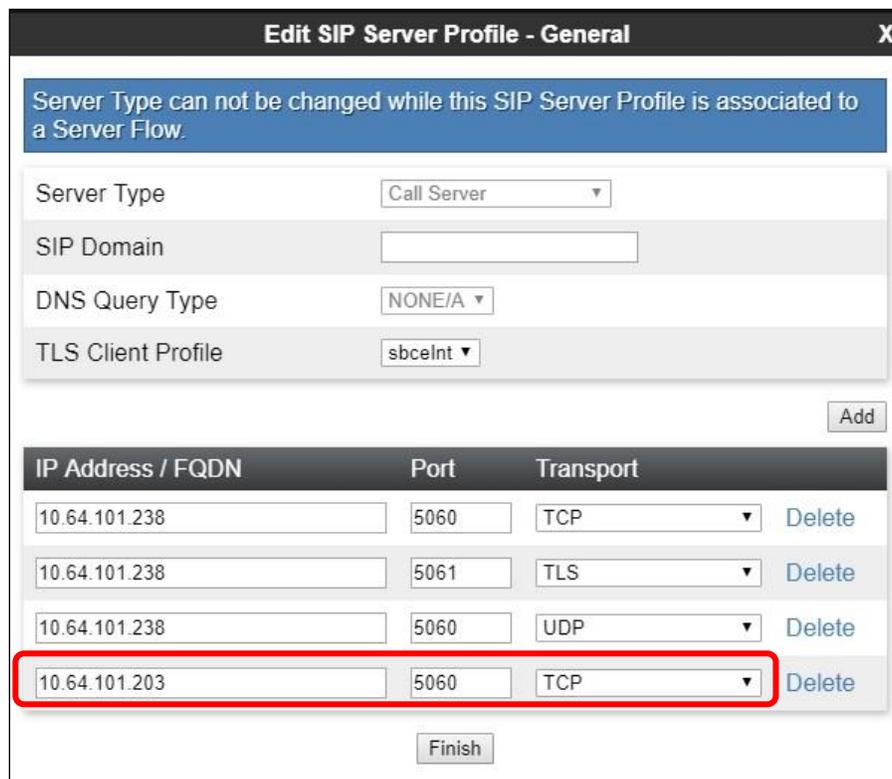
Server Type	Call Server	
TLS Client Profile	sbcelnt	
DNS Query Type	NONE/A	
IP Address / FQDN	Port	Transport
10.64.101.238	5060	TCP
10.64.101.238	5061	TLS
10.64.101.238	5060	UDP

An 'Edit' button is located at the bottom right of the configuration table, highlighted with a red box.

The **Edit SIP Server Profile – General** pop-up screen is displayed. Click **Add** to add an entry.



In the new entry, enter the IP address of the Voice Screening Proxy server for **IP Address / FQDN**. For **Port** and **Transport**, enter and select the values correspond to the Voice Spam Filter SIP entity link in **Section 5.2.1**.



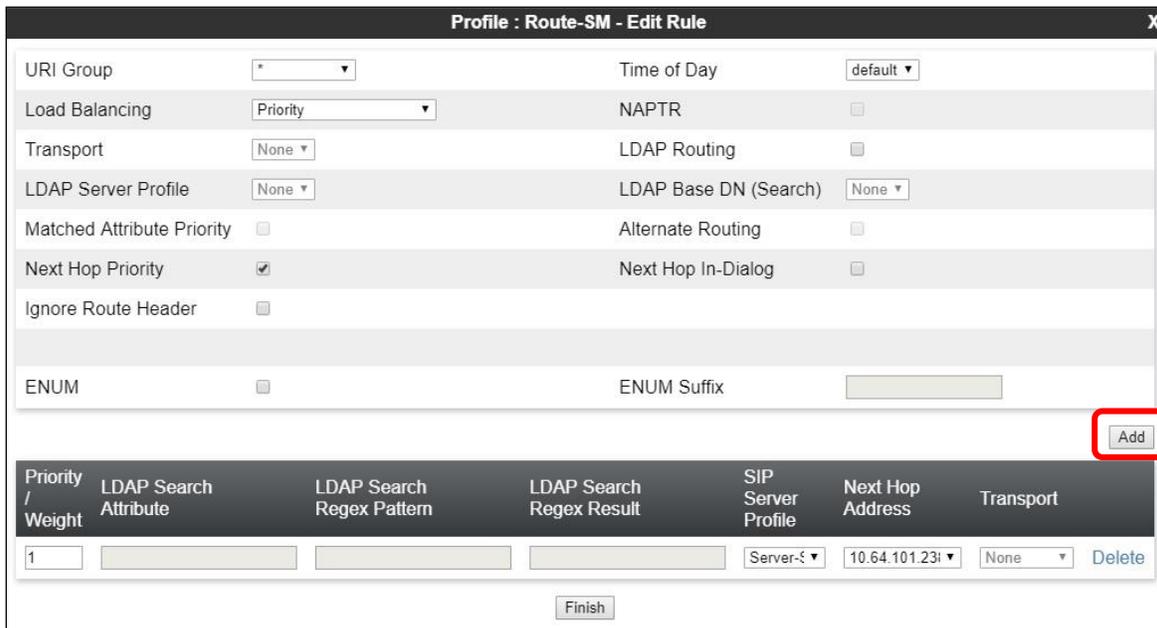
6.3. Administer Routing Profile

Select **Backup/Restore** → **Configuration Profiles** → **Routing** from the left pane to display the existing routing profiles.

Select the routing profile associated with Session Manager, in this case “Route-SM”, as shown below. Click **Edit**.



The **Profile : Route-SM – Edit Rule** pop-up screen is displayed. Click **Add** to add an entry.



In the existing entry, update the **Priority / Weight** to a lesser priority, such as “2” as shown below.

In the new entry, enter the following values for the specified fields and retain the default values for the remaining fields.

- **Priority / Weight:** The highest priority of “1”.
- **SIP Server Profile:** The SIP server profile for Session Manager, in this case “Server-SM”.
- **Next Hop Address:** Select the address entry associated with Voice Screening Proxy.

With this routing configuration, inbound calls to be routed from SBCE to Session Manager will now route to Voice Screening Proxy as primary and will only route to Session Manager as alternate when the Voice Screening Proxy is not available.

Profile : Route-SM - Edit Rule

URI Group: * Time of Day: default

Load Balancing: Priority NAPTR:

Transport: None LDAP Routing:

LDAP Server Profile: None LDAP Base DN (Search): None

Matched Attribute Priority: Alternate Routing:

Next Hop Priority: Next Hop In-Dialog:

Ignore Route Header:

ENUM: ENUM Suffix:

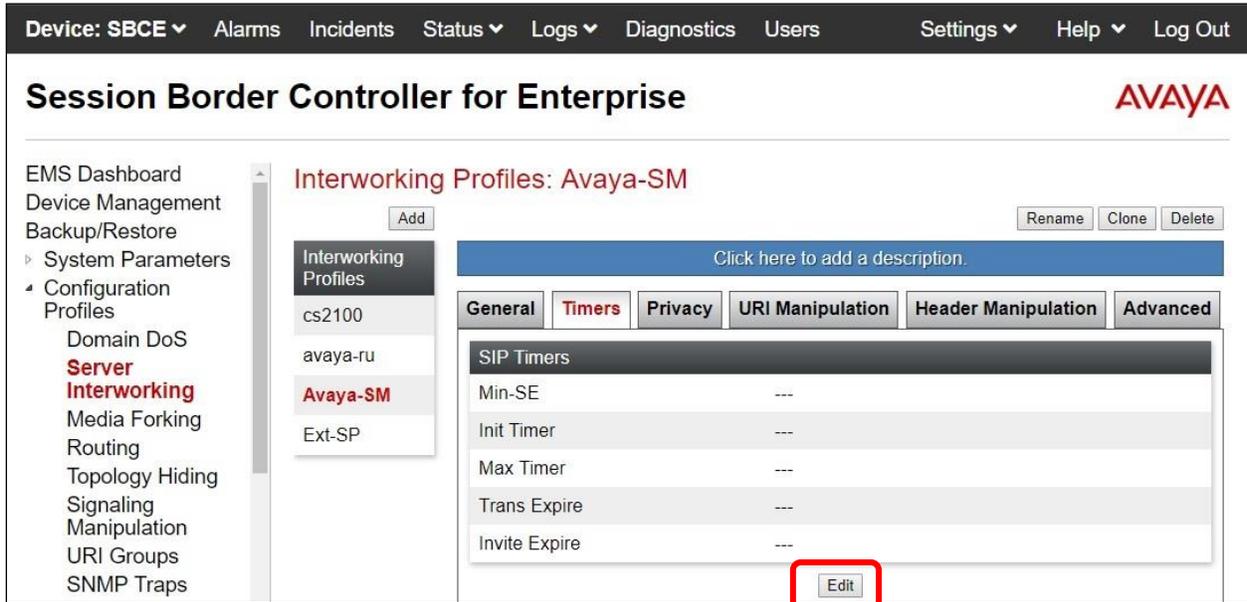
Add

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	
2	<input type="text"/>	<input type="text"/>	<input type="text"/>	Server-S	10.64.101.231	None	Delete
1	<input type="text"/>	<input type="text"/>	<input type="text"/>	Server-S	10.64.101.203:5060 (TCP)	None	Delete

Finish

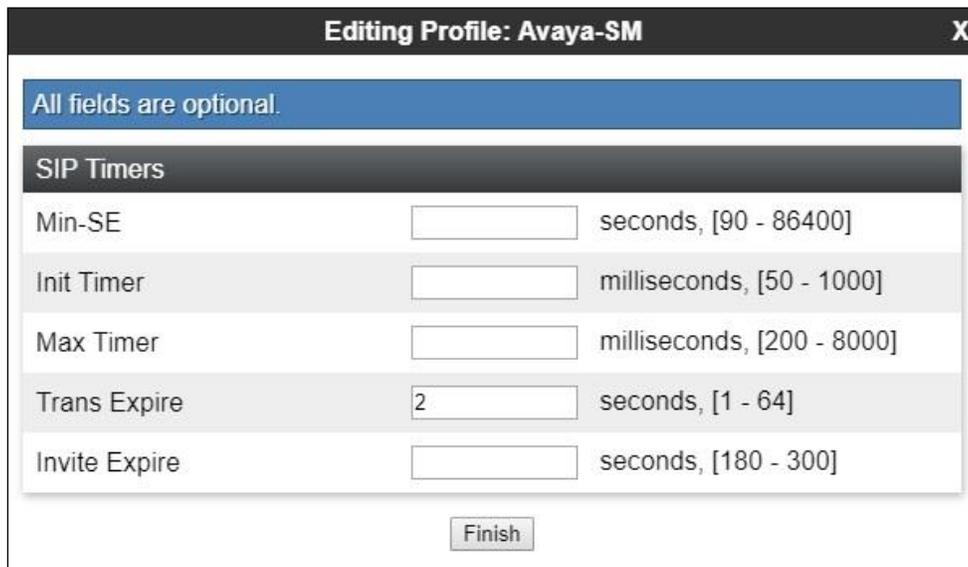
6.4. Administer Interworking Profile

Select **Backup/Restore** → **Configuration Profiles** → **Server Interworking** from the left pane to display the existing interworking profiles. Select the interworking profile associated with Session Manager, in this case “Avaya-SM”, as shown below. Select the **Timers** tab in the right pane and click **Edit**.



The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Device: SBCE', 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Session Border Controller for Enterprise' with the AVAYA logo. The left sidebar contains a navigation menu with 'Server Interworking' selected. The main content area is titled 'Interworking Profiles: Avaya-SM' and features a list of profiles: 'cs2100', 'avaya-ru', 'Avaya-SM', and 'Ext-SP'. The 'Avaya-SM' profile is selected. Below the list, there are tabs for 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'Timers' tab is active, displaying a table of SIP Timers with the following values: Min-SE (---), Init Timer (---), Max Timer (---), Trans Expire (---), and Invite Expire (---). An 'Edit' button is highlighted with a red box at the bottom right of the table.

The **Editing Profile: Avaya-SM** pop-up screen is displayed. For **Trans Expire**, enter an appropriate short duration. In the compliance testing, two seconds was used as the allotted time for SBCE to wait for a route response from Voice Screening Proxy as primary before routing to Session Manager as alternate.



The screenshot shows the 'Editing Profile: Avaya-SM' pop-up screen. The title bar reads 'Editing Profile: Avaya-SM' with a close button (X). Below the title bar, there is a blue bar with the text 'All fields are optional.' The main content area is titled 'SIP Timers' and contains a table with the following fields and values:

Timer	Value	Unit	Range
Min-SE	<input type="text"/>	seconds	[90 - 86400]
Init Timer	<input type="text"/>	milliseconds	[50 - 1000]
Max Timer	<input type="text"/>	milliseconds	[200 - 8000]
Trans Expire	2	seconds	[1 - 64]
Invite Expire	<input type="text"/>	seconds	[180 - 300]

A 'Finish' button is located at the bottom center of the screen.

7. Configure Mutare Voice Spam Filter

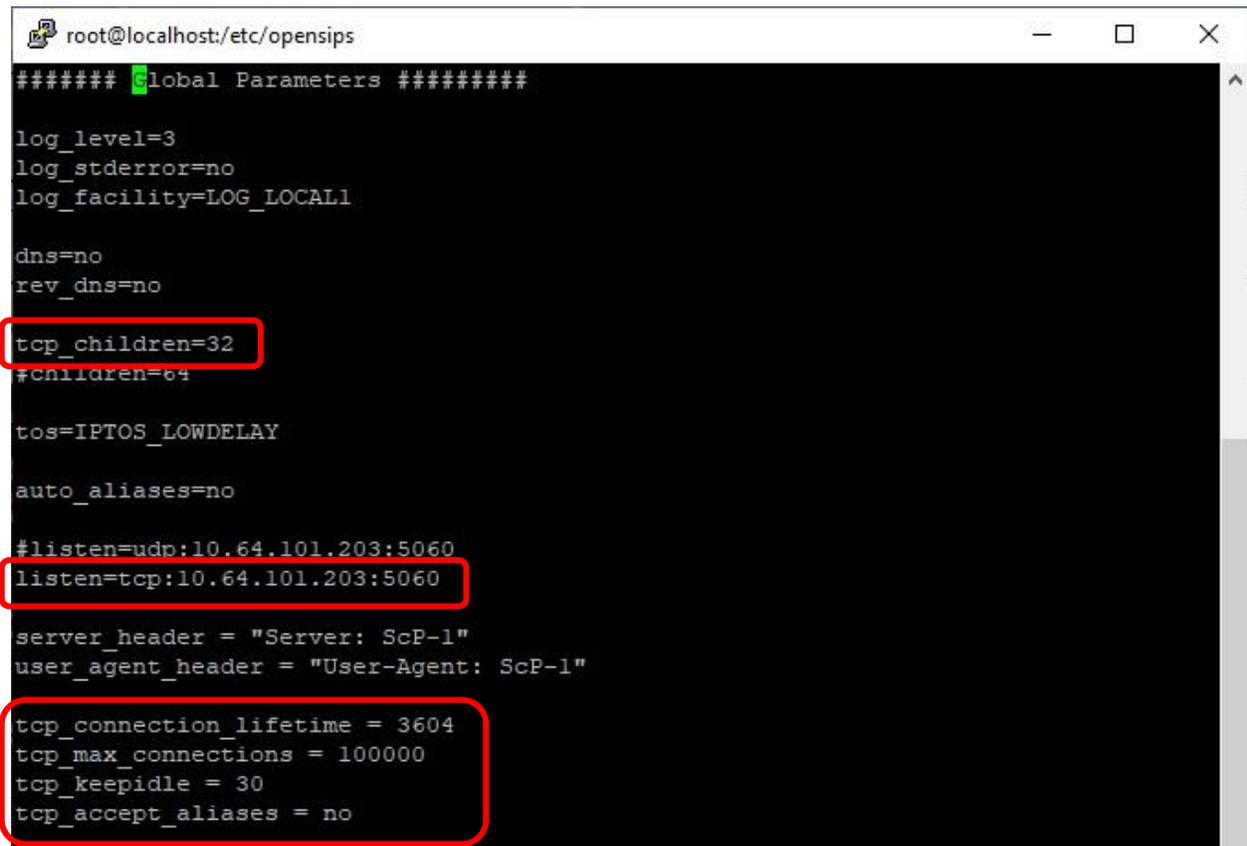
This section provides the procedures for configuring Voice Spam Filter. The procedures include the following areas:

- Administer opensips.cfg
- Administer SQL
- Administer control panel
- Administer rules manager

The configuration of Voice Spam Filter is typically performed by Mutare operations technician. The procedural steps are presented in these Application Notes for information purposes. This section assumes that values for API URL, Connect URL, appliance ID, account ID, and token have all been obtained from Voice Application Server and configured on Voice Screening Proxy.

7.1. Administer opensips.cfg

Log in to the Linux shell of the Voice Screening Proxy server with super user credentials. Navigate to the `/etc/opensips` directory and edit the `opensips.cfg` file. Scroll down to the **Global Parameters** sub-section and uncomment out 6 TCP related parameters shown below. For the **listen** parameter, replace the default IP address with the IP address of the Voice Screening Proxy server.



```
root@localhost:/etc/opensips
##### Global Parameters #####
log_level=3
log_stderr=no
log_facility=LOG_LOCAL1

dns=no
rev_dns=no
tcp_children=32
#children=64

tos=IPTOS_LOWDELAY

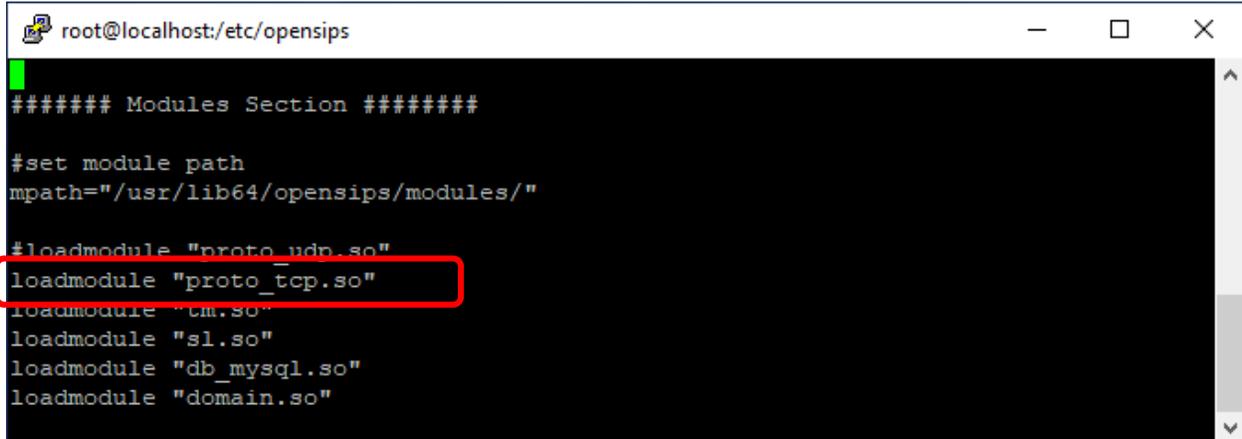
auto_aliases=no

#listen=udp:10.64.101.203:5060
listen=tcp:10.64.101.203:5060

server_header = "Server: ScP-1"
user_agent_header = "User-Agent: ScP-1"

tcp_connection_lifetime = 3604
tcp_max_connections = 100000
tcp_keepidle = 30
tcp_accept_aliases = no
```

Scroll down to the **Modules Section** and uncomment out the TCP related module shown below.

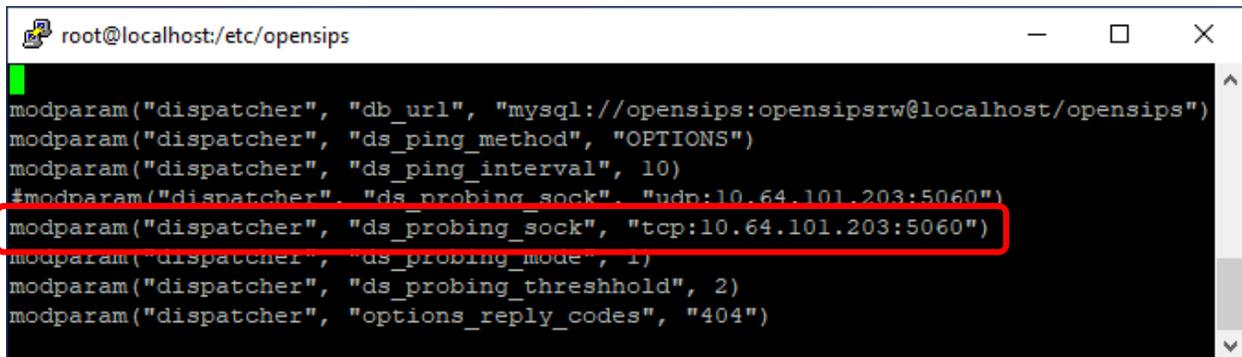


```
root@localhost:/etc/opensips
##### Modules Section #####

#set module path
mpath="/usr/lib64/opensips/modules/"

#loadmodule "proto_udp.so"
loadmodule "proto_tcp.so"
loadmodule "tm.so"
loadmodule "sl.so"
loadmodule "db_mysql.so"
loadmodule "domain.so"
```

Scroll down to the section shown below, uncomment out the TCP related line and replace the default IP address with the IP address of Voice Screening Proxy as shown below.



```
root@localhost:/etc/opensips

modparam("dispatcher", "db_url", "mysql://opensips:opensipsw@localhost/opensips")
modparam("dispatcher", "ds_ping_method", "OPTIONS")
modparam("dispatcher", "ds_ping_interval", 10)
#modparam("dispatcher", "ds_probing_sock", "udp:10.64.101.203:5060")
modparam("dispatcher", "ds_probing_sock", "tcp:10.64.101.203:5060")
modparam("dispatcher", "ds_probing_mode", 1)
modparam("dispatcher", "ds_probing_threshold", 2)
modparam("dispatcher", "options_reply_codes", "404")
```

Scroll down to the **route [resume]** sub-section and replace the default IP address with the Session Manager signaling IP address in the highlighted area shown below. This setting will use Session Manager as the next hop.

```
@localhost:/etc/opensips
route [resume] {
    xlog("L_INFO","Locust API response: return code = $rc, HTTP code = $var(rcode), body =
    $var(body)\n");
    $var(rc) = $rc;
    ## we suppose api returns 1 in $var(body) if the Caller ID is blacklisted
    if ($var(rc) == "1" && $var(rcode) == "200" && $var(body) == '{"status":"drop"}') {
        xlog("L_INFO","Call from $fU to $tU is denied because $fU is blacklisted\n");
        send_reply("403","Forbidden");
    } else if ($var(rc) == "1" && $var(rcode) == "200" && $var(body) =~ "refer.*") {
        xlog("L_INFO","Call from $fU to $tU is being redirected to $var(body)\n");
        $avp(newuri) = $(var(body){s.select,2,:});
        $var(reg) = '/"/g';
        $var(reg_1) = '/)/';
        $avp(newuri_1) = $(avp(newuri){re.subst,$var(reg)});
        $avp(newuri_2) = $(avp(newuri_1){re.subst,$var(reg_1)});
        $ru = "sip:" + $avp(newuri_2);
        xlog("L_INFO","R-URI is now $ru\n");
        if ($rd != "10.64.101.238") {
            route($ru);
            xlog("L_INFO","REDIRECTED TO $ru\n");
            exit;
        }
    }
}
```

7.2. Administer SQL

From the command line, enter the two SQL commands shown below to update the next hop destination to the IP address of the Session Manager signaling interface.

```
root@localhost:/home/mutareadmin
@localhost mutareadmin]# mysql -uopensips -popensipsrw
Warning: Using a password on the command line interface can be insecure.
Welcome to the MySQL monitor.  Commands end with ; or \g.
Your MySQL connection id is 561
Server version: 5.6.44 MySQL Community Server (GPL)

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Oracle is a registered trademark of Oracle Corporation and/or its
affiliates. Other names may be trademarks of their respective
owners.

Type 'help;' or '\h' for help. Type '\c' to clear the current input statement.

mysql> UPDATE opensips.dispatcher set destination='sip:10.64.101.238:5060' where id=1;
Query OK, 1 row affected (0.01 sec)
Rows matched: 1  Changed: 1  Warnings: 0

mysql>
```

From the command line, enter the first SQL command below to set the TCP socket, and the second SQL command below to make certain the TCP socket has been set correctly.

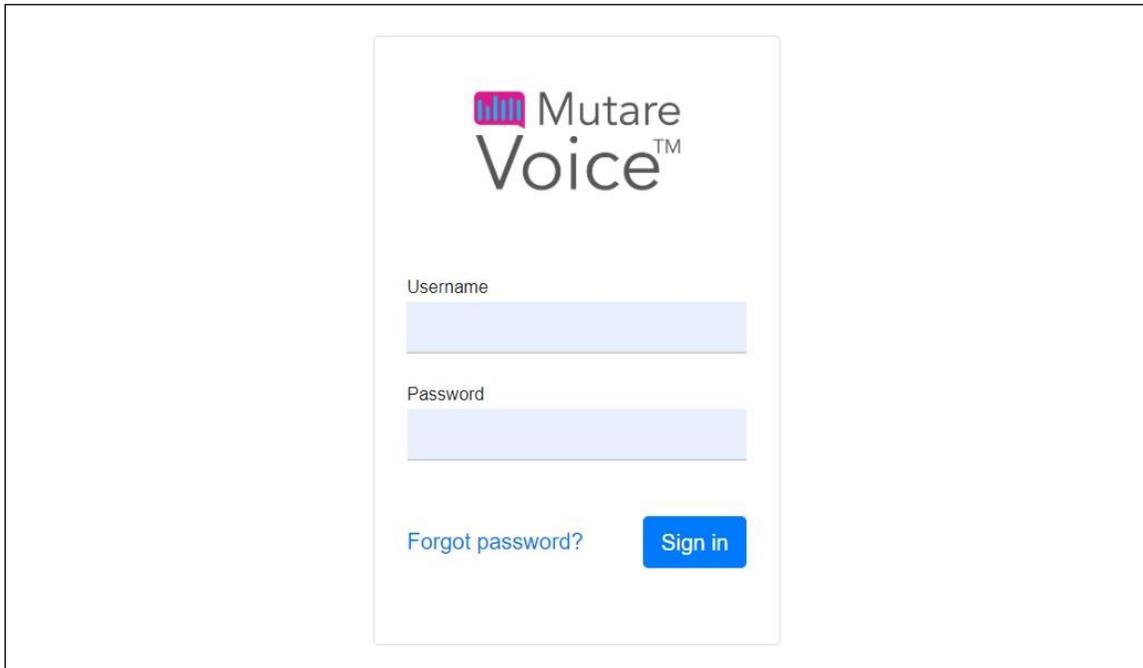
```
root@localhost:/home/mutareadmin
mysql>
mysql> update opensips.dispatcher set socket='tcp:10.64.101.203:5060' where setid=1;
Query OK, 1 row affected (0.00 sec)
Rows matched: 1 Changed: 1 Warnings: 0

mysql>
mysql> select * from opensips.dispatcher;
+-----+-----+-----+-----+-----+-----+-----+-----+
| id | setid | destination | socket | state | weight | priority | attr |
+-----+-----+-----+-----+-----+-----+-----+-----+
| 1 | 1 | sip:10.64.101.238:5060 | tcp:10.64.101.203:5060 | 0 | 1 | | 0 |
| PBX | | | | | | | |
+-----+-----+-----+-----+-----+-----+-----+-----+
1 row in set (0.00 sec)

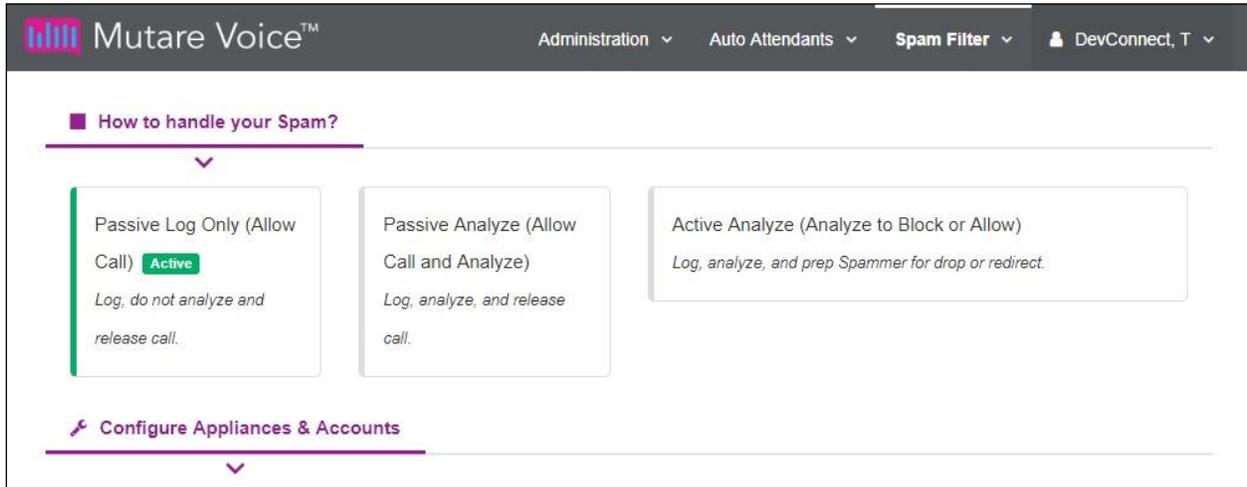
mysql>
```

7.3. Administer Control Panel

Access the Voice Spam Filter web interface by using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the Voice Application Server. The screen below is displayed. Log in using the appropriate credentials.

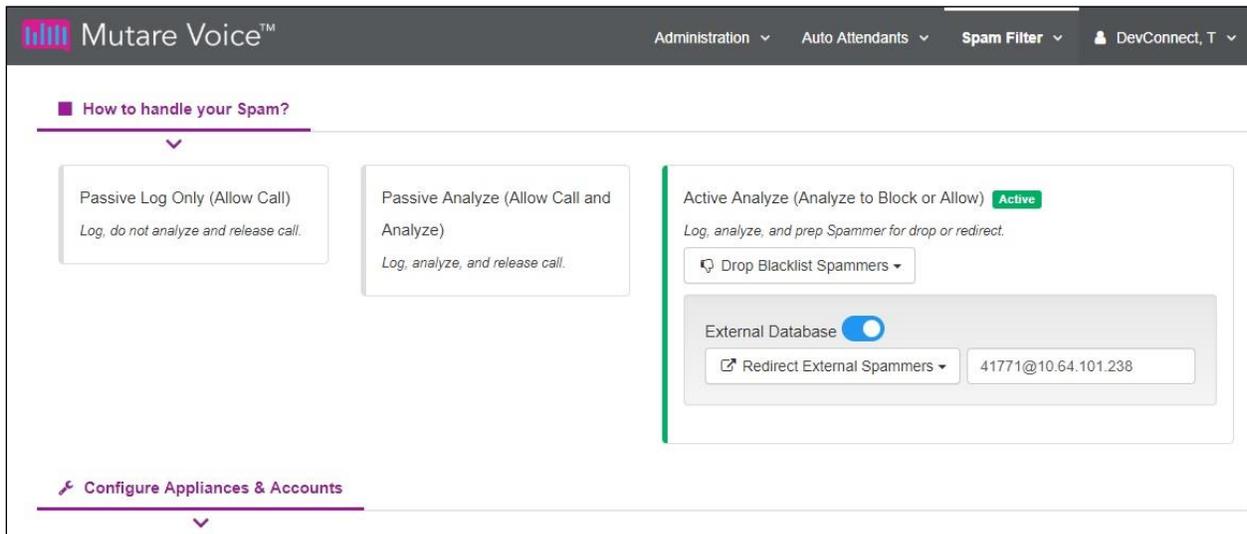


In the subsequent screen (not shown), select **Spam Filter** → **Control Panel** from the top menu to display the screen below.



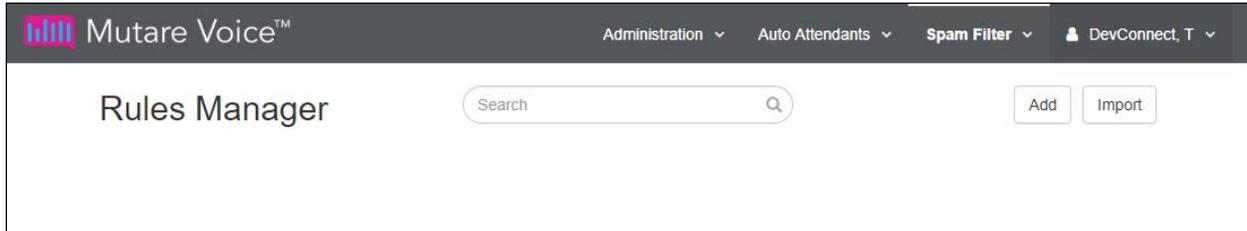
Follow reference [4] to configure the desired action for handling of spam calls. The screenshot below shows a sample configuration with all calls to be analyzed, calls from calling parties on the enterprise blacklist to be dropped, and calls from calling parties on the robocall external database to be redirected.

For redirected calls, enter “x@y” as destination where “x” is a desired resource extension and “y” is the signaling IP address of Session Manager. In the compliance testing, “41771” corresponded to an announcement extension on Communication Manager.

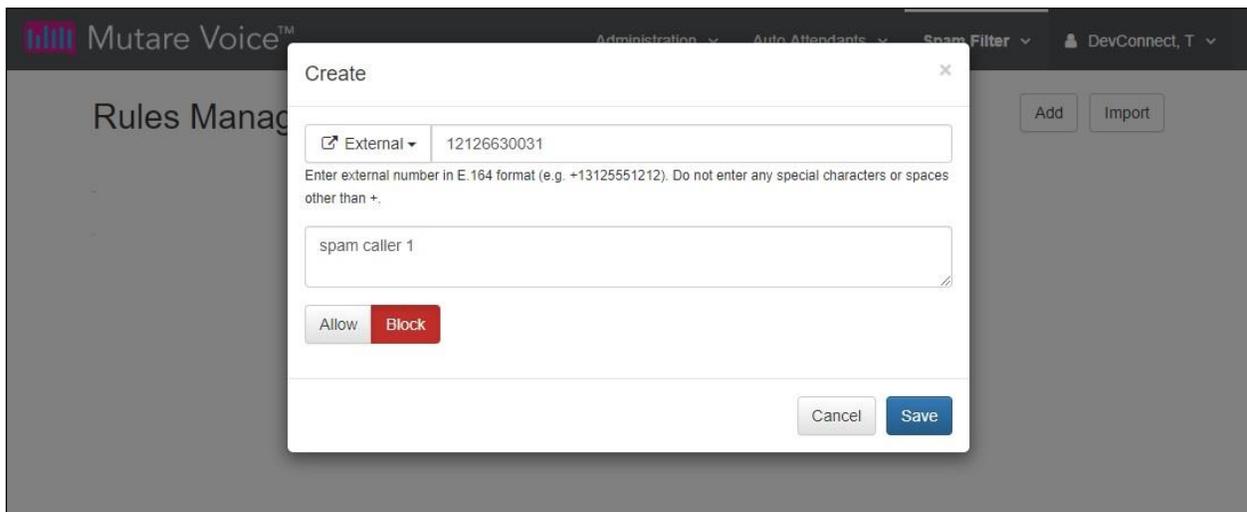


7.4. Administer Rules Manager

Select **Spam Filter** → **Rules Manager** from the top menu to display the **Rules Manager** screen below. Click **Import** to import a CSV file with existing numbers or **Add** to add individual numbers. In the compliance testing, **Add** was used.

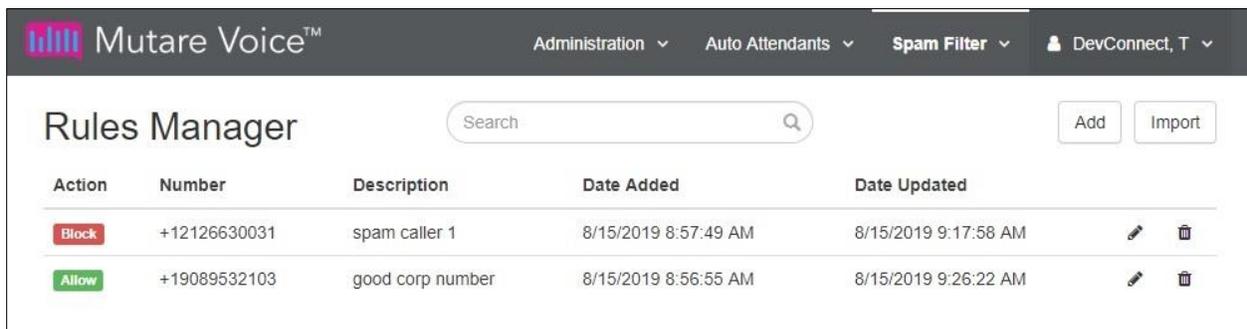


The **Create** pop-up box is displayed next. Enter a ten-digits calling number preceded with “1”, a brief description, and select **Allow** for whitelist or **Block** for blacklist.



Repeat the procedures in this section to configure all calling numbers for the enterprise whitelist and blacklist.

In the compliance testing, two entries were created as shown below. Note that Voice Spam Filter automatically converted the numbers into E.164 format by adding the plus sign.



Action	Number	Description	Date Added	Date Updated		
Block	+12126630031	spam caller 1	8/15/2019 8:57:49 AM	8/15/2019 9:17:58 AM		
Allow	+19089532103	good corp number	8/15/2019 8:56:55 AM	8/15/2019 9:26:22 AM		

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager, SBCE, and Voice Spam Filter.

8.1. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements** → **Session Manager** from the top menu to display the **Session Manager Dashboard** screen (not shown).

Select **Session Manager** → **System Status** → **SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click on the Voice Spam Filter entity name from **Section 5.2.1**.

AVAYA Aura® System Manager 8.1

Users | Elements | Services | Widgets | Shortcuts | Search | Help ?

Home | Session Manager x

Session Manager

- Dashboard
- Session Manager Ad...
- Global Settings
- Communication Prof...
- Network Configur...
- Device and Locati...
- Application Conf...
- System Status
- SIP Entity Monit...**
- Managed Band...
- Security Module...
- SIP Firewall Stat...
- Registration Su...
- User Registration

SIP Entity Link Monitoring Status Summary

This page provides a summary of Session Manager SIP entity link monitoring status.

SIP Entities Status for All Monitoring Session Manager Instances

Run Monitor As of 1:46 PM

1 Item Filter: Enable

Session Manager	Type	Monitored Entities					Total
		Down	Partially Up	Up	Not Monitored	Deny	
<input type="checkbox"/> DR-SM	Core	2	0	7	0	0	9

Select : All, None

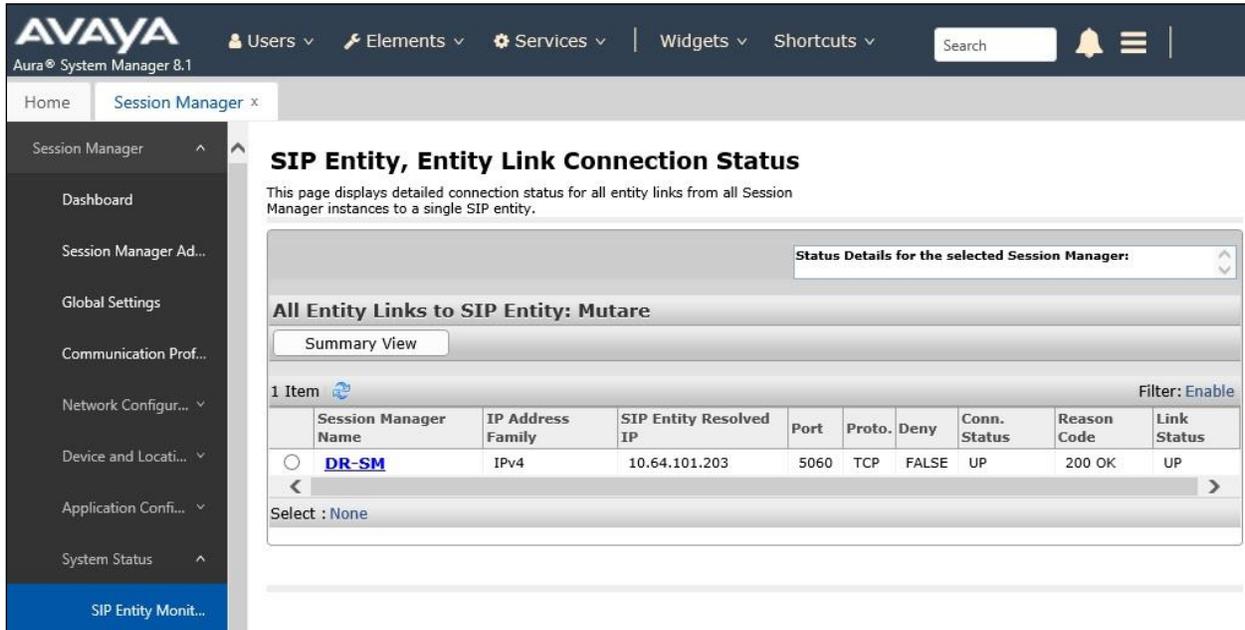
All Monitored SIP Entities

Run Monitor

9 Items Filter: Enable

SIP Entity Name
<input type="checkbox"/> ACCS1-IP500V2
<input type="checkbox"/> IPO1-IP500V2
<input type="checkbox"/> IPO2-IP500V2
<input type="checkbox"/> DR-MSG
<input type="checkbox"/> DR-CM
<input type="checkbox"/> IPO2-IPOSE
<input type="checkbox"/> SBCE
<input type="checkbox"/> DR-CM-5212
<input checked="" type="checkbox"/> Mutare

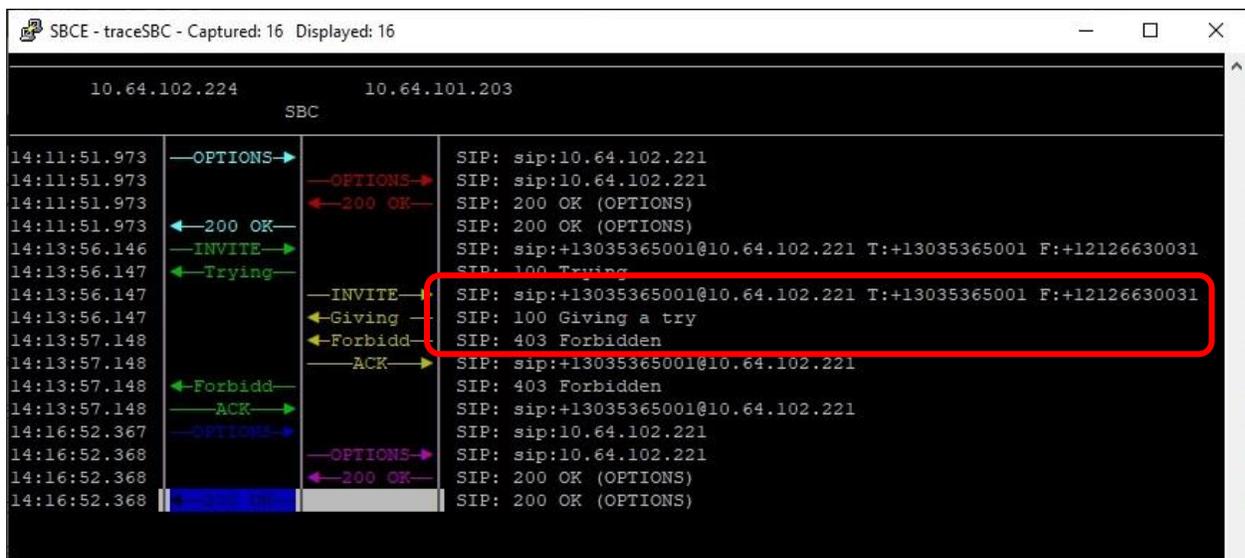
The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are “UP”, as shown below.



8.2. Verify Avaya Session Border Controller for Enterprise

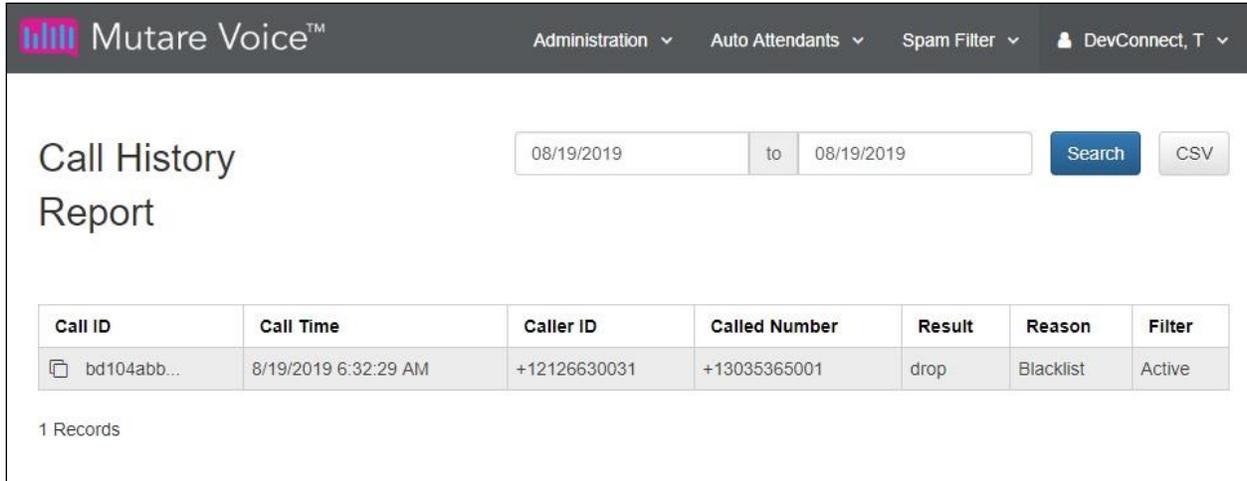
Log in to the Linux shell of the SBCE management interface with appropriate credentials and run the “tracesbc” command.

Make an inbound call from a PSTN caller with calling number on the enterprise blacklist from **Section 7.4**. Verify that the SBCE trace shows a **403 Forbidden** response from Voice Screening Proxy, and that the PSTN caller receives a call rejection treatment from the SIP Service Provider.



8.3. Verify Mutare Voice Spam Filter

From the Voice Spam Filter web interface, select **Spam Filter** → **Call History** from the top menu. Verify that there is an entry associated with the last call along with appropriate **Result** and **Reason** as shown below.



The screenshot shows the Mutare Voice web interface. At the top, there is a navigation bar with the Mutare Voice logo and several menu items: Administration, Auto Attendants, Spam Filter, and a user profile for DevConnect, T. Below the navigation bar, the main content area is titled "Call History Report". There is a search filter section with two date input fields, both containing "08/19/2019", a "to" separator, a "Search" button, and a "CSV" button. Below the search section is a table with the following data:

Call ID	Call Time	Caller ID	Called Number	Result	Reason	Filter
 bd104abb...	8/19/2019 6:32:29 AM	+12126630031	+13035365001	drop	Blacklist	Active

Below the table, it indicates "1 Records".

9. Conclusion

These Application Notes describe the configuration steps required for Mutare Voice Spam Filter to successfully interoperate with Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

This section references the product documentation relevant to these Application Notes.

1. *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 3, August 2019, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019, available at <http://support.avaya.com>.
3. *Administering Avaya Session Border Controller for Enterprise*, Release 8.0.x, Issue 4, August 2019, available at <http://support.avaya.com>.
4. *Mutare Voice Admin Guide*, Version 1.9.0, June 26, 2019, available at <https://mutare.com/knowledge/tech-docs>.

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