DevConnect Program

Application Notes for Configuring Cox Communications SIP Trunking with Avaya IP Office 12.0 and Avaya Session Border Controller 10.2 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking service between service provider Cox Communications and Avaya IP Office Release 12.0 and Avaya Session Border Controller Release 10.2.

Cox Communications SIP Trunk Service provides PSTN access via a SIP trunk between the enterprise and the Cox Communications network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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

Cox Communications is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the Avaya DevConnect Program.

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1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking service between Cox Communications and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of Avaya IP Office Server Edition Release 12.0, Avaya IPO Voicemail Pro, Avaya Workplace for Windows (SIP mode), Avaya H.323, Avaya SIP, digital and analog deskphones. The enterprise solution connects to the Cox Communications network via the Avaya Session Border Controller (Avaya SBC). Cox Managed CPE (Edgewater EdgeMarc 2900E SIP Application-Layer Gateway) is included as part of the Service Provider service and not as part of the CPE solution (See **Section 11 Appendix** for more information).

The Cox Communications referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office connecting to Cox Communications via the Avaya SBC.

This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**. **Note**: NAT devices added between Avaya SBC and the Cox Communications network should be transparent to the SIP signaling.

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 this DevConnect Application Note 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.

2.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office and Avaya SBC was connected to Cox Communications. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, digital, and analog phones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider
- Outgoing PSTN calls from various phone types. Phone types included H.323, SIP, digital, and analog phones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider
- Inbound and outbound PSTN calls from/to the Avaya Workplace for Windows (SIP)
- Inbound and outbound long hold time call stability
- Various call types including: local, long distance, international call, inbound toll-free, outbound toll-free, outbound calls to Assisted Operator, 411 Local Directory Assistance call, 911 Emergency call during the compliance testing
- SIP transport UDP/RTP and Port 5060 between Cox Communications and the simulated Avaya enterprise site
- Codec G.711MU
- Caller number/ID presentation
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls
- DTMF transmission using RFC 2833
- SIP OPTIONS queries and responses
- Voicemail navigation for inbound and outbound calls
- Telephony features such as hold and resume, transfer, and conference
- T.38 fax and G.711 pass-through mode
- Off-net call forwarding
- Off-net call transfer
- Twinning to mobile phones on inbound calls
- SIP Trunk registration between Avaya SBC and Cox Managed CPE
- Remote Worker. Avaya Workplace for Windows (SIP) was used to test remote worker functionality. Note: Remote Worker was tested as part of this solution. The configuration necessary to support remote worker is beyond the scope of these Application Notes and are not included in these Application Notes. For these configuration details, see **Reference [8] in Section 10**.

Item not supported include the following:

- TLS/SRTP SIP transport
- Call redirection to the PSTN using the SIP REFER method was not tested during the compliance test, Cox Communications did not fully support it. Cox Communications confirmed that SIP REFER is only supported without sending a NOTIFY message. Therefore, the SIP RE-INVITE method for call redirection to the PSTN was used instead.

2.2. Test Results

Interoperability testing of Cox Communications was completed with successful results for all test cases with the exception of the observation described below:

- The Cox Managed CPE equipment did not forward SIP Diversion headers (or PAI headers) to the Cox Communications network during call forward scenarios to the PSTN. This behaviour had no negative impact on the forwarded calls. It is being mentioned here simply as an observation. This issue should be fixed in a future firmware release for the Cox Communications Managed CPE equipment residing at the enterprise.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit: http://support.avaya.com.

For technical support on Cox Communications SIP Trunking, contact Cox Communications at http://www.cox.com.

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to Cox Communications through the public internet. For confidentiality and privacy purposes, actual public IP addresses and DID numbers used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

The Avaya components used to create the simulated customer site included:

- IP Office Server Edition Primary Server
- IP Office Voicemail Pro
- IP Office Server Edition Expansion System (IP500 V2)
- Avaya Session Border Controller
- Avaya 96x1 Series IP Deskphones (H.323)
- Avaya 11x0 Series IP Deskphones (SIP)
- Avaya J129 IP Deskphones (SIP)
- Avaya 1408 Digital phones
- Avaya Analog phones
- Avaya Workplace for Windows (SIP)
- Avaya Workplace for Windows (SIP) for remote worker.

The Primary Server consists of a Dell PowerEdge R640 server, running the Avaya IP Office Server Edition Linux software Release 12.0. Avaya Voicemail Pro runs as a service on the Primary Server. The LAN1 port of Avaya IP Office is connected to Avaya SBC internal interface. The Avaya SBC external interface is connected to Cox Managed CPE LAN interface while Cox Managed CPE WAN interface is connected to Cox Communications' network via public network.

The optional Expansion System (IP500 V2) is used for the support of digital, analog, fax, and additional IP stations. It consists of an Avaya IP Office IP500 V2 with the MOD DGTL STA16 expansion module which provides connections for 16 digital stations to the PSTN, and the extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs

A separate Windows 10 Enterprise PC runs Avaya IP Office Manager to configure and administer Avaya IP Office system.

Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user's phones will also ring and can be answered at configured mobile phones.

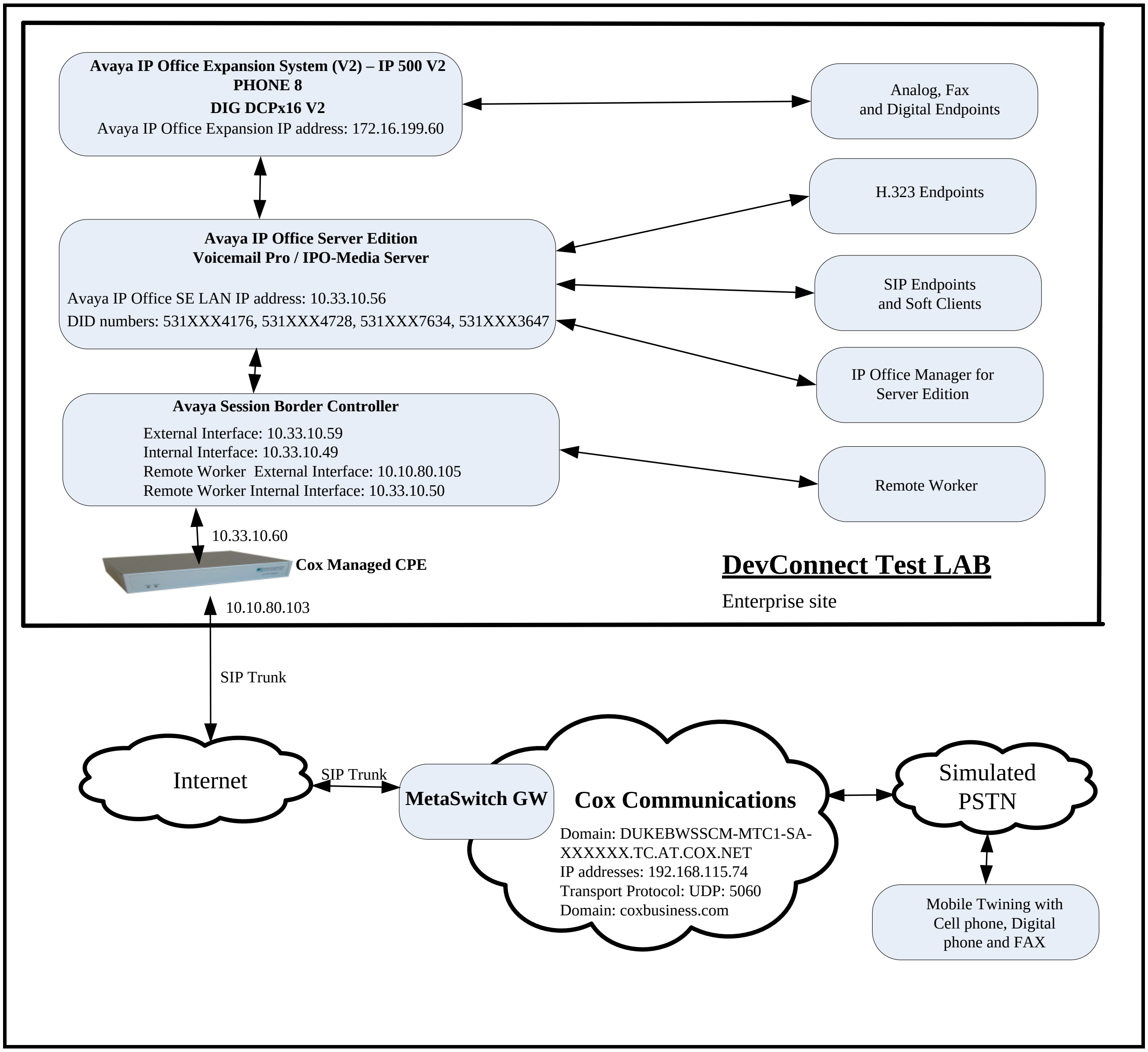


Figure 1 - Test Configuration for Avaya IP Office with Cox Communications SIP Trunk Service

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 9 + N digits to send digits across the SIP trunk to Cox Communications. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Cox Communications. For the compliance test, outbound calls to Canadian numbers within the North American Numbering Plan (NANP) were tested. The user would dial 11 (1 + 10) digits. For these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the To field of an outbound SIP INVITE message, and it was configured to send 10 digits in the From field. For inbound calls, Cox Communications sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and Avaya SBC, such as a data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and SRTP traffic between the service provider and Avaya SBC must be allowed to pass through these devices.

4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya IP Office Server Edition solution ▪ Primary Server Dell PowerEdge R640 – IPO-Linux-PC ▪ IPO-Media Server ▪ Voicemail Pro ▪ IP Office Manager for Server Edition ▪ IP Office Expansion System (V2) – IP 500 V2 ▪ IP Office Analogue - PHONE 8 ▪ IP Office Digital - DIG DCPx16 V2	 12.0.0.0.0 build 56 12.0.0.0.0 build 56 12.0.0.0.0 build 26 12.0.0.0.0 build 56 12.0.0.0.0 build 56 12.0.0.0.0 build 56 12.0.0.0.0 build 56
Avaya Session Border Controller running on VMware®-based Avaya appliance	10.2.0.0-86-24077
Avaya 1140E IP Deskphone (SIP)	04.04.33
Avaya 9641G IP Deskphone (H323)	6.8.5.5.1
Avaya 9621G IP Deskphone (H323)	6.8.5.5.1
Avaya J129 IP Deskphone (SIP)	4.0.7.1.5
Avaya Workplace Client for Windows	3.37.0.156.28
Avaya 1408D Digital Deskphone	R48
Avaya Analog Deskphone	N/A
VentaFax	7.10.258.664
Cox Communications	
Cox Managed CPE	16.4.0.1
MetaSwitch GW	4.3.40

5. Configure Avaya IP Office Solution

This section describes the Avaya IP Office Server Edition solution configuration necessary to support connectivity to the Cox Communications via Avaya SBC. It is assumed that the initial installation and provisioning of the Server Edition Primary Server and Expansion System has been previously completed and therefore is not covered in these Application Notes. For information on these installation tasks refer to the Additional References **Section 10**.

This section describes the Avaya IP Office Server Edition configuration to support connectivity to Cox Communications system via Avaya SBC. Avaya IP Office Server Edition is configured through the Avaya IP Office Server Edition Manager PC application. From a PC running the Avaya IP Office Server Edition Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office Server Edition system from the pop-up window. Log in using appropriate credentials.

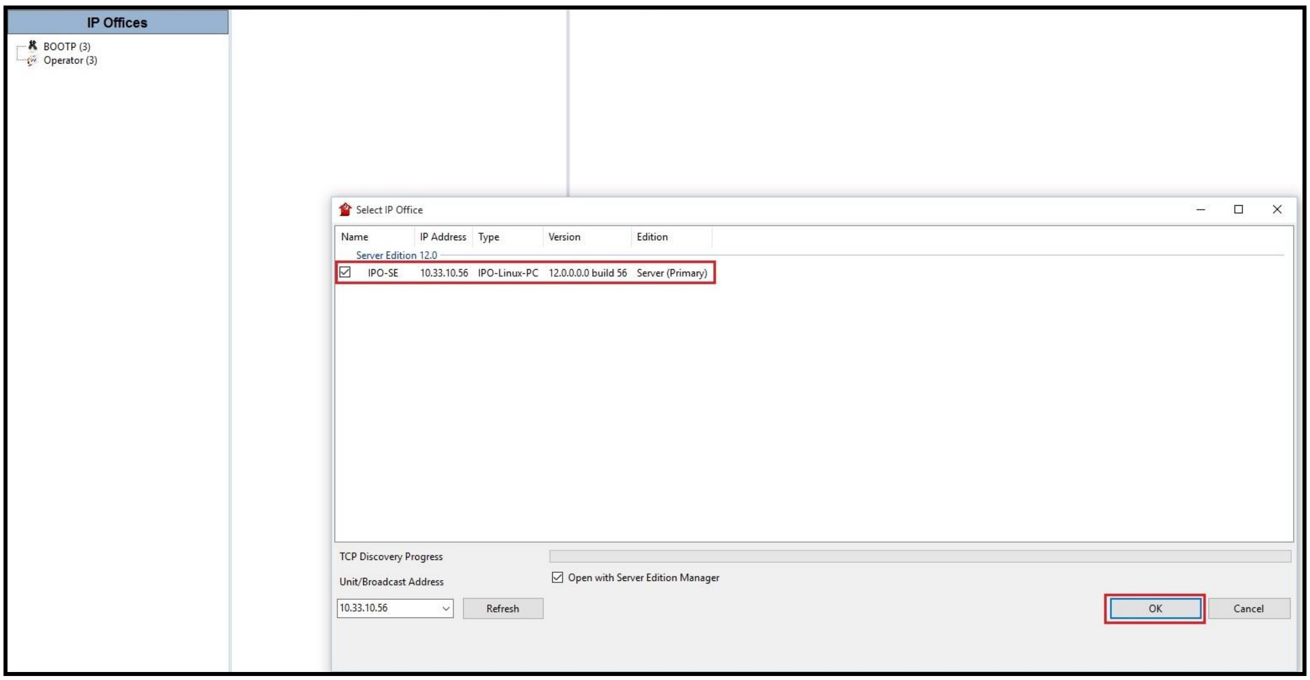


Figure 2 – Avaya IP Office Server Edition Selection

The appearance of the Avaya IP Office Server Edition Manager can be customized using the **View** menu. In the screens presented in this section, it includes the system inventory of the servers and links for administration and configuration tasks.

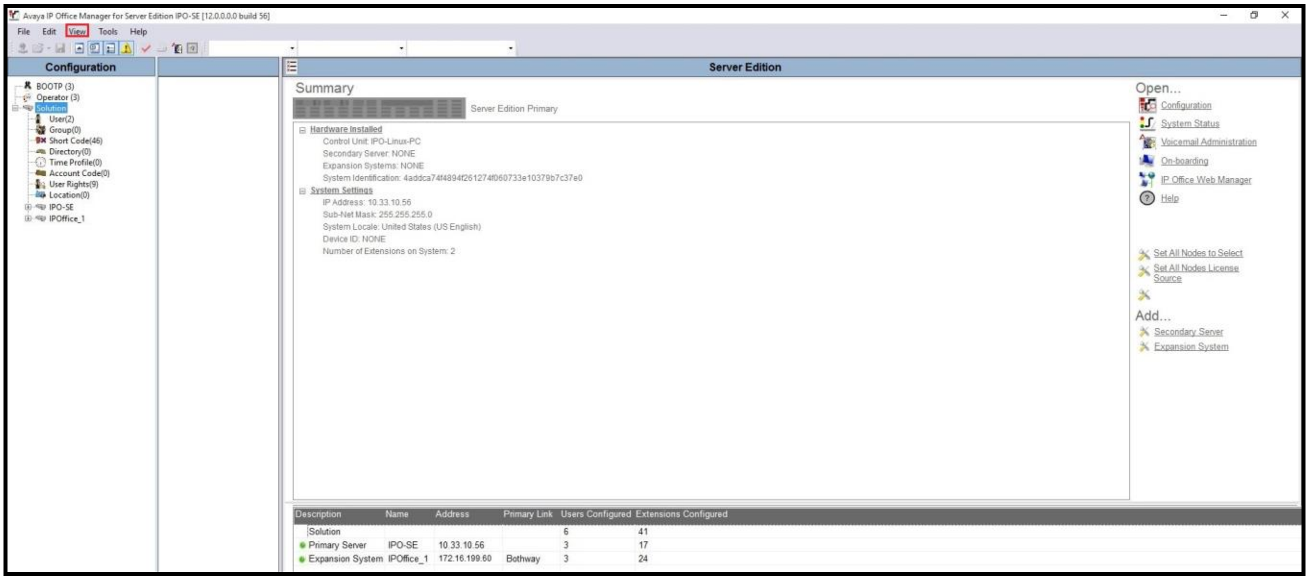
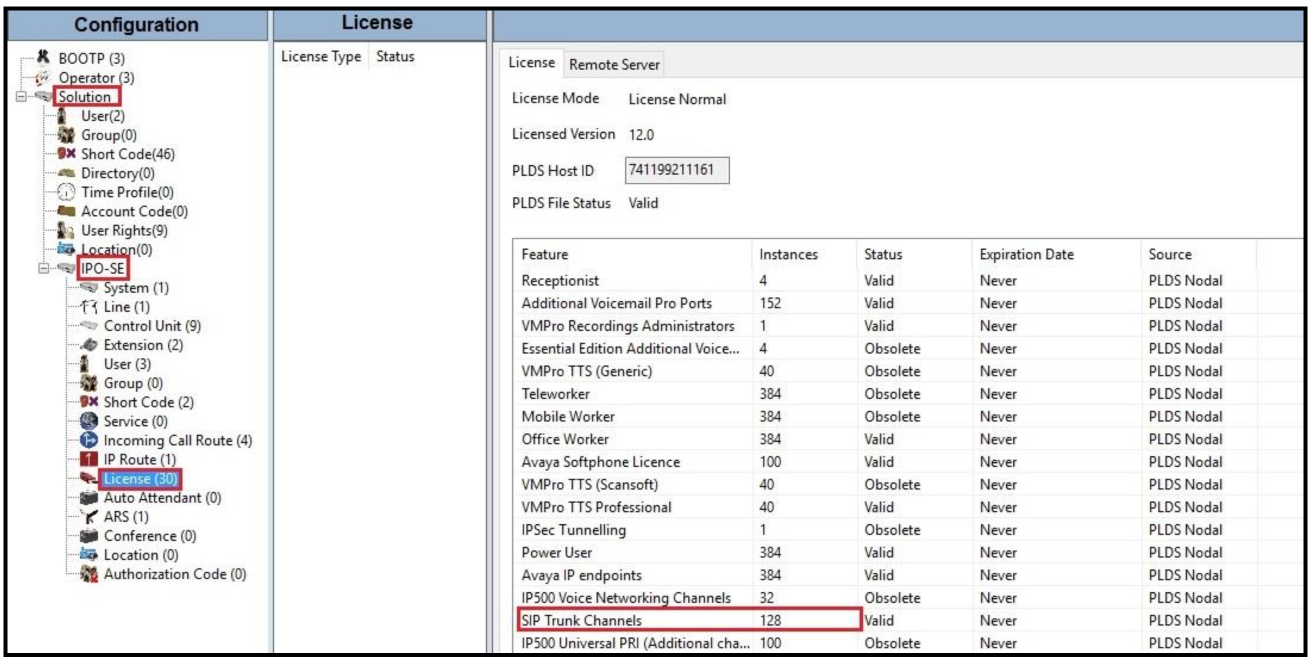


Figure 3 – Avaya IP Office Server Edition View Menu

5.1. Licensing

The configuration and features described in these Application Notes require the Avaya IP Office Server Edition system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

Licenses for an Avaya IP Office Server Edition solution are based on a combination of centralized licensing done through the Avaya IP Office Server Edition Primary Server, and server specific licenses that are entered into the configuration of the system requiring the feature. SIP Trunk Channels are centralized licenses, and they are entered into the configuration of the Primary Server. Note that when centralized licenses are used to enable features on other systems, such as SIP trunk channels, the Primary Server allocates those licenses to the other systems only after it has met its own license needs. To verify that there is a SIP Trunk Channels license with sufficient capacity, select **Solution → IPO-SE → License** on the Navigation pane and SIP Trunk Channels in the Group pane. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane.



Feature	Instances	Status	Expiration Date	Source
Receptionist	4	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	152	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Essential Edition Additional Voice...	4	Obsolete	Never	PLDS Nodal
VMPro TTS (Generic)	40	Obsolete	Never	PLDS Nodal
Teleworker	384	Obsolete	Never	PLDS Nodal
Mobile Worker	384	Obsolete	Never	PLDS Nodal
Office Worker	384	Valid	Never	PLDS Nodal
Avaya Softphone Licence	100	Valid	Never	PLDS Nodal
VMPro TTS (Scansoft)	40	Obsolete	Never	PLDS Nodal
VMPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Power User	384	Valid	Never	PLDS Nodal
Avaya IP endpoints	384	Valid	Never	PLDS Nodal
IP500 Voice Networking Channels	32	Obsolete	Never	PLDS Nodal
SIP Trunk Channels	128	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal

Figure 4 – Avaya IP Office Server Edition License

5.2. TLS Management

For the compliance test, the signaling on the SIP trunk between IP Office and the Avaya SBC was secured using TLS. Testing was done using identity certificates signed by a local certificate authority, Avaya Aura® System Manager. The generation and installation of these certificates are beyond the scope of these Application Notes. However, once the certificates are available, they can be viewed on IP Office in the following manner.

To view the certificates currently installed on IP Office, navigate to **File → Advanced → Security Settings**. Log in with the appropriate security credentials (not shown). In the Security Settings window, navigate **to Security → System** and select the **Certificates** tab.

To verify the identity certificate, locate the **Identity Certificate** section and click **View** to see the details of the certificate.
To verify the trusted certificate, locate the **Trusted Certificate Store** section, select the trusted certificate and click **View** to see the details of it.

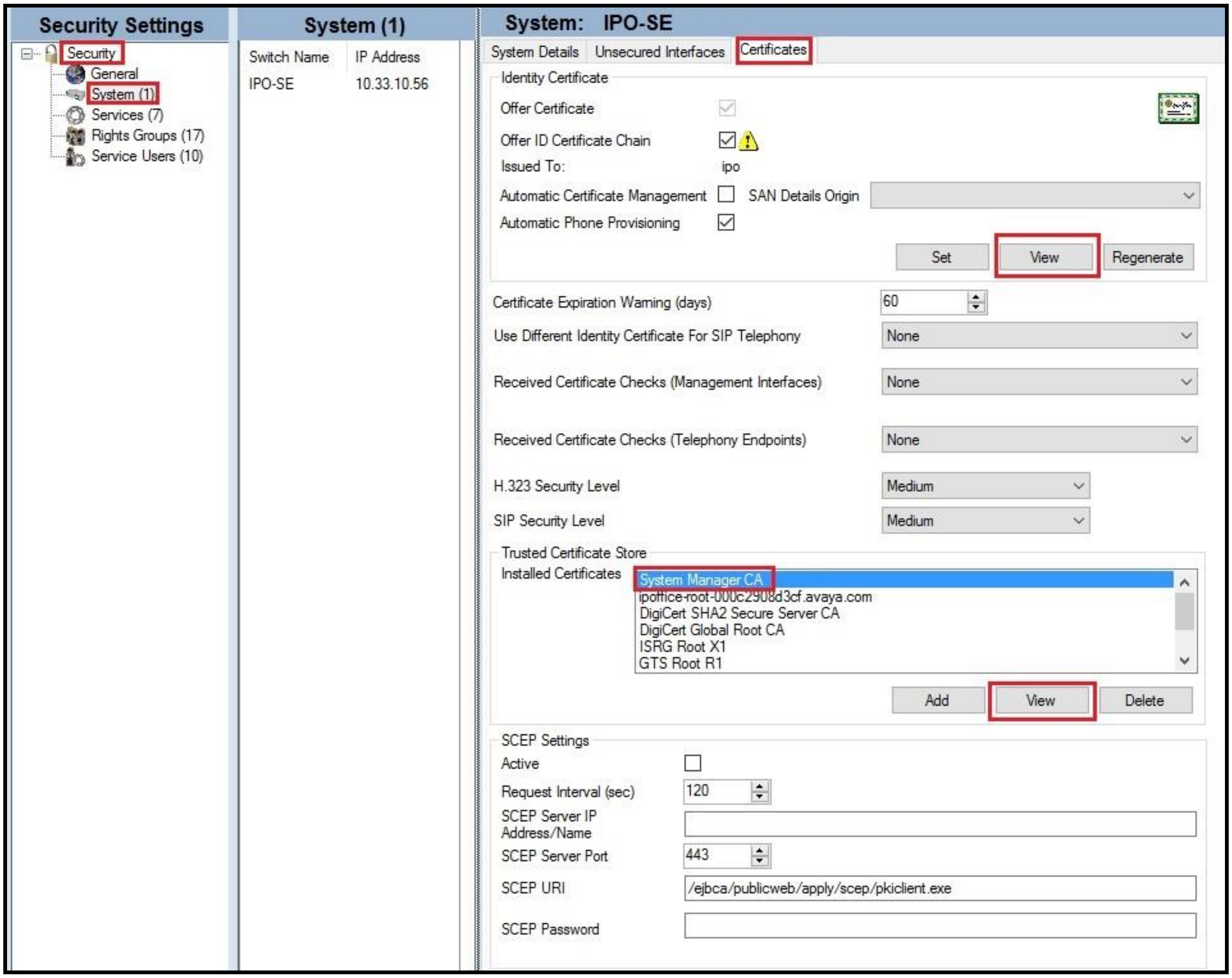


Figure 5 – Avaya IP Office Server Edition TLS Certificate

5.3. System Settings

Configure the necessary system settings. In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side). For the compliance test, the **LAN1** interface was used to connect IP Office to the enterprise private network (LAN), **LAN2** was not used.

5.3.1. System – LAN1 Tab

In the sample configuration, **IPO-SE** was used as the Primary Server name and **IPOffice_1** was used as the Expansion System name. The **LAN1** port on the Primary Server (Eth0) connects to the inside interface (enterprise private network side) of the Avaya SBC across the enterprise LAN (private) network. The LAN1 port on the Expansion System were used to connect to the enterprise LAN (private) network. The outside interface of the Avaya SBC connects to Cox Communications network via the public internet.

To configure the LAN1 settings on the Primary Server, complete the following steps. Navigate to **IPO-SE → System (1)** in the Navigation and Group Panes and then navigate to the **LAN1 → LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office Server Edition LAN1 port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements. Click **OK** to submit the change.

Figure 6 - Avaya IP Office Primary Server LAN1 Settings

The **VoIP** tab as shown in the screenshot below was configured with following settings:

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Deskphones/Softphones using the H.323 protocol to register
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to Avaya SBC
- Check the **SIP Register Enable** to allow Avaya IP deskphones/softphones to register using the SIP protocol
- Input **SIP Domain Name** and **SIP Register FQDN** as **10.33.10.56**
- The **Layer 4 Protocol** uses **TLS** with **TLS Port** as **5061**
- Verify **Keepalives** to select **Scope** as **RTP-RTCP** with **Periodic timeout 60** and select **Initial keepalives** as **Enabled**
- All other parameters should be set according to customer requirements
- Click **OK** to submit the changes

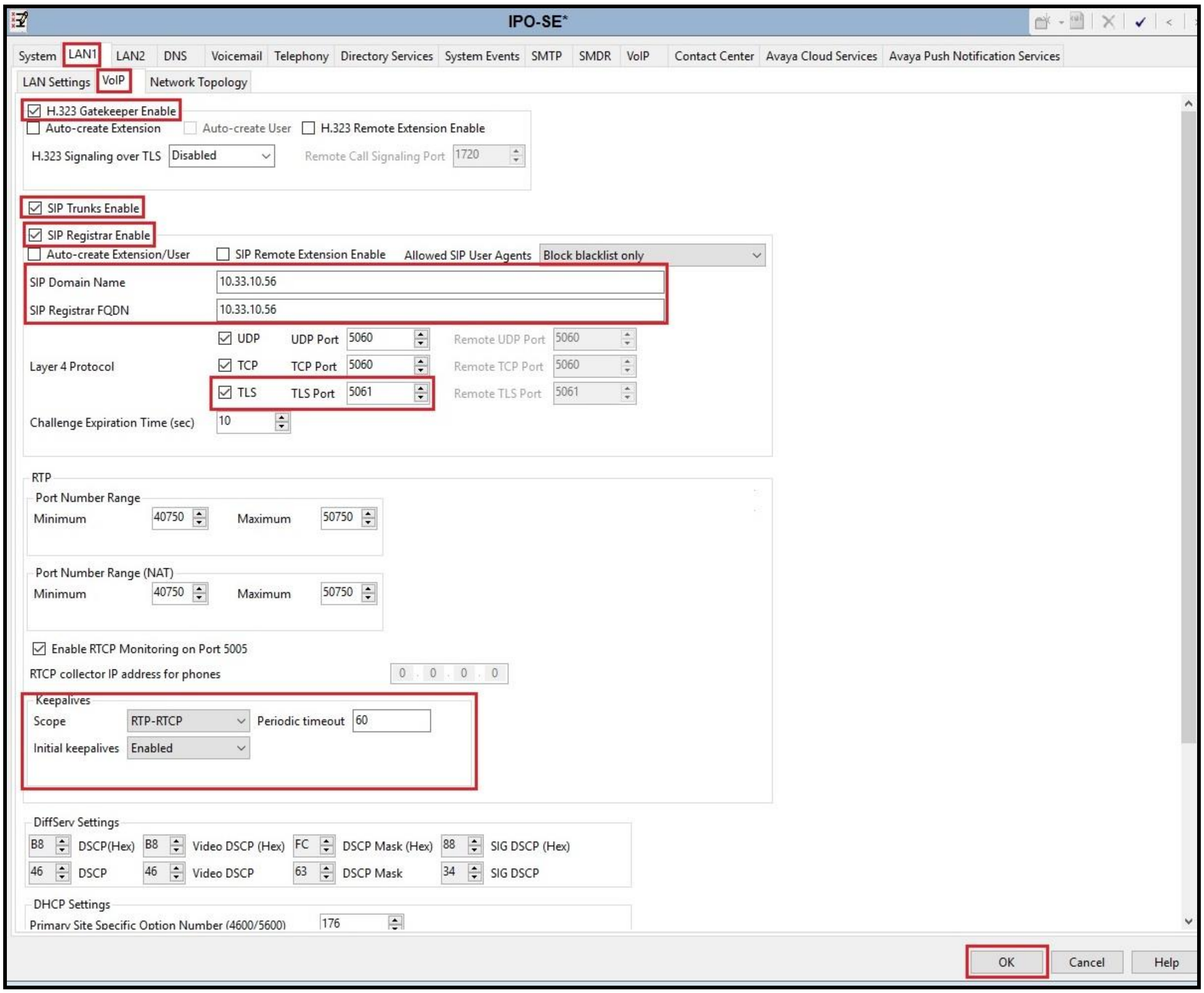


Figure 7 - Avaya IP Office Primary Server LAN1 VoIP

To configure the LAN1 settings tab for the Expansion System, navigate to **Solution → IPOffice_1 → System (1)** in the Navigation and Group Panes and then navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields should be populated with the values assigned during the Expansion System initial installation process. Verify the configuration or modify the values if needed. While DHCP was disabled during the compliance test, this parameter should be set according to customer requirements. Other settings were left at their default values. Click **OK** to submit the change.

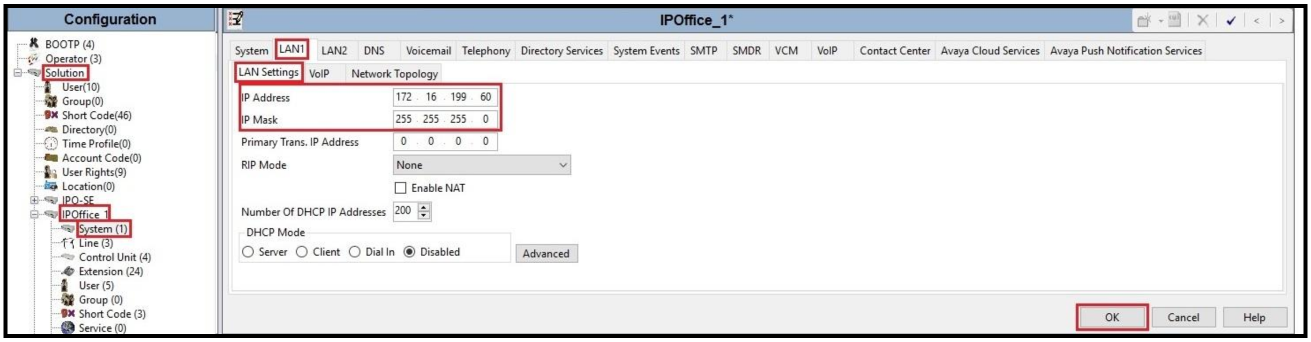


Figure 8 - Avaya IP Office Expansion Server LAN Settings

5.3.2. System – Telephony Tab

Navigate to **Solution → IPO-SE → System (1)** in the Navigation and Group Panes (not shown) and then navigate to the **Telephony → Telephony** tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. **U-Law** is used for Switch and Line. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the service provider across the SIP trunk. The **Hold Timeout (sec)** field controls how long calls remain on hold before being alerted to the user and should be set based on the customer's requirement. Set **Default Name Priority** to **Favor Trunk** to have IP Office display the name provided in the Caller ID from the SIP trunk. Defaults were used for all other settings. Click **OK** to submit the changes.

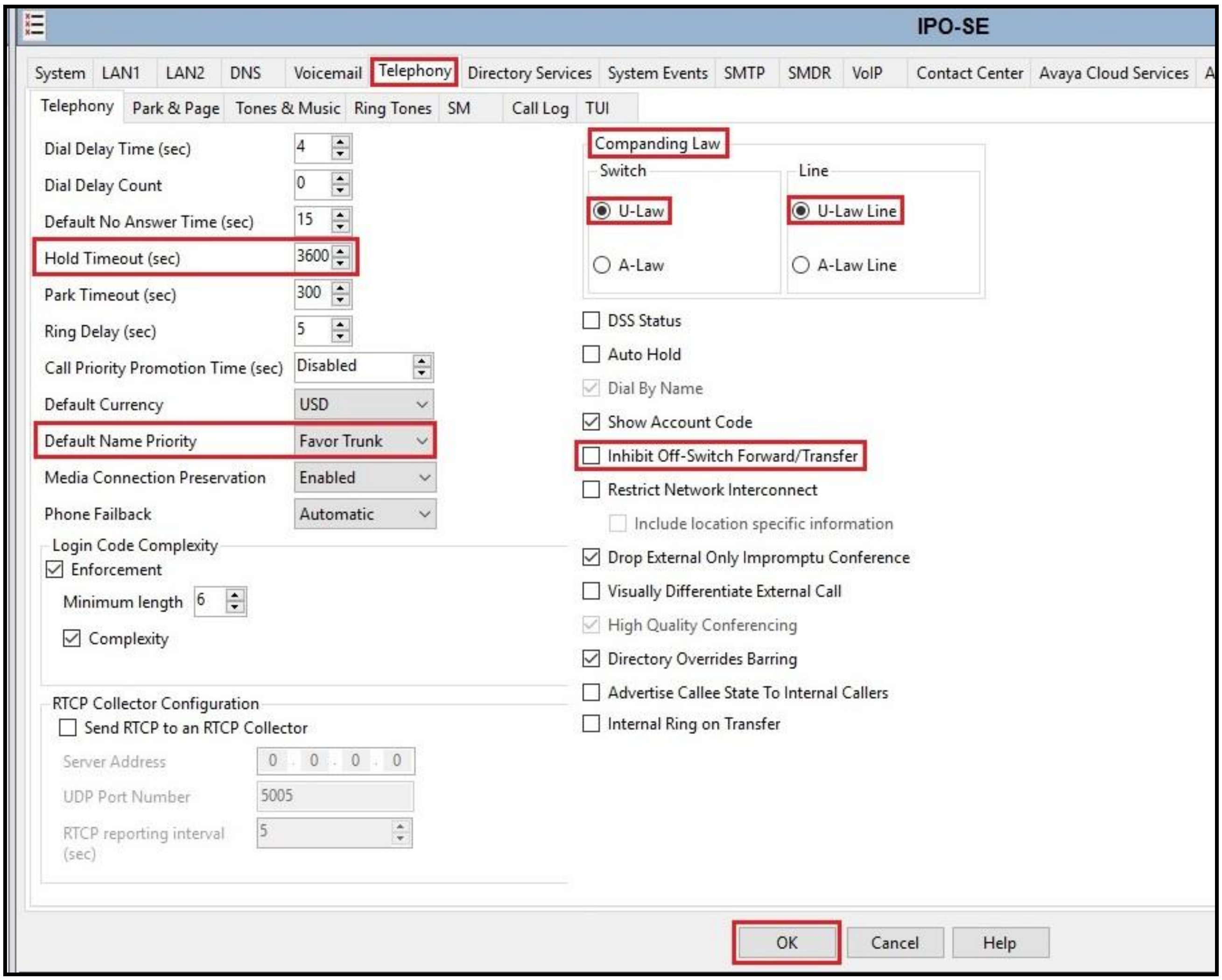


Figure 9 - Avaya IP Office Primary Server Telephony

Navigate to **Solution → IPOffice_1 → System (1)** (not shown) and repeat the steps above to configure the **Telephony** settings for the Expansion System.

5.3.3. System – VoIP Tab

Navigate to the **VoIP** tab in the Details pane to view or change the system codecs and VoIP security settings.

Navigate to **Solution → IPO-SE → System (1)** in the Navigation and Group Panes and then navigate to the **VoIP** tab in the Details Pane. Leave the **RFC2833 Default Payload** as the default value of **101**. The buttons between the two lists can be used to move codecs between the **Unused** and **Selected** lists, and to change the order of the codecs in the **Selected** codecs list. By default, all IP lines and phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific line or extension. The example below shows the codecs used for IP phones (SIP and H.323), the system's default codecs and order were used.
Click **OK** to submit the changes.

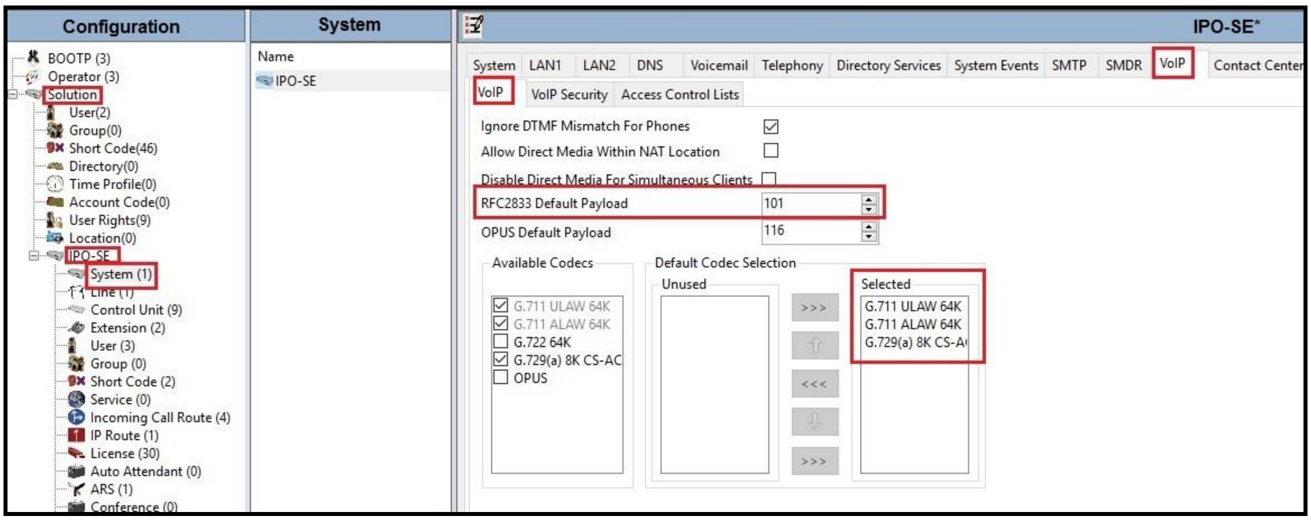


Figure 10 - Avaya IP Office Primary Server VoIP

Note: The codec selections defined under this section (VoIP – VoIP tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 5.5.2** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

Secure Real-Time Transport Protocol (SRTP) refers to the application of additional encryption and or authentication to VoIP calls (SIP and H.323). SRTP can be applied between telephones, between ends of an IP trunk or in various other combinations.

Configuring the use of SRTP at the system level is done on the **VoIP Security** tab using the Media Security setting. The options are:

- Disabled (default)
- Preferred
- Enforced

When enabling SRTP on the system, the recommended setting is **Preferred**. In this scenario, IP Office uses SRTP if supported by the far-end, otherwise uses RTP. If the **Enforced** setting is used, and SRTP is not supported by the far-end, the call is not established.

To configure the use of SRTP, navigate to **Solution → IPO-SE → System (1)** in the Navigation and Group Panes and then navigate to **VoIP → VoIP Security** tab on the Details pane.
Set the **Media Security** drop-down menu to **Preferred** to have IP Office attempt use encrypted RTP for devices that support it and fall back to RTP for devices that do not support encryption.

- Verify **Strict SIPS** is not checked
- Under **Media Security Options**, select **RTP** for the **Encryptions** and **Authentication** fields
- Under **Crypto Suites**, select **SRTP_AES_CM_128_SHA1_80**
- Click **OK** to commit

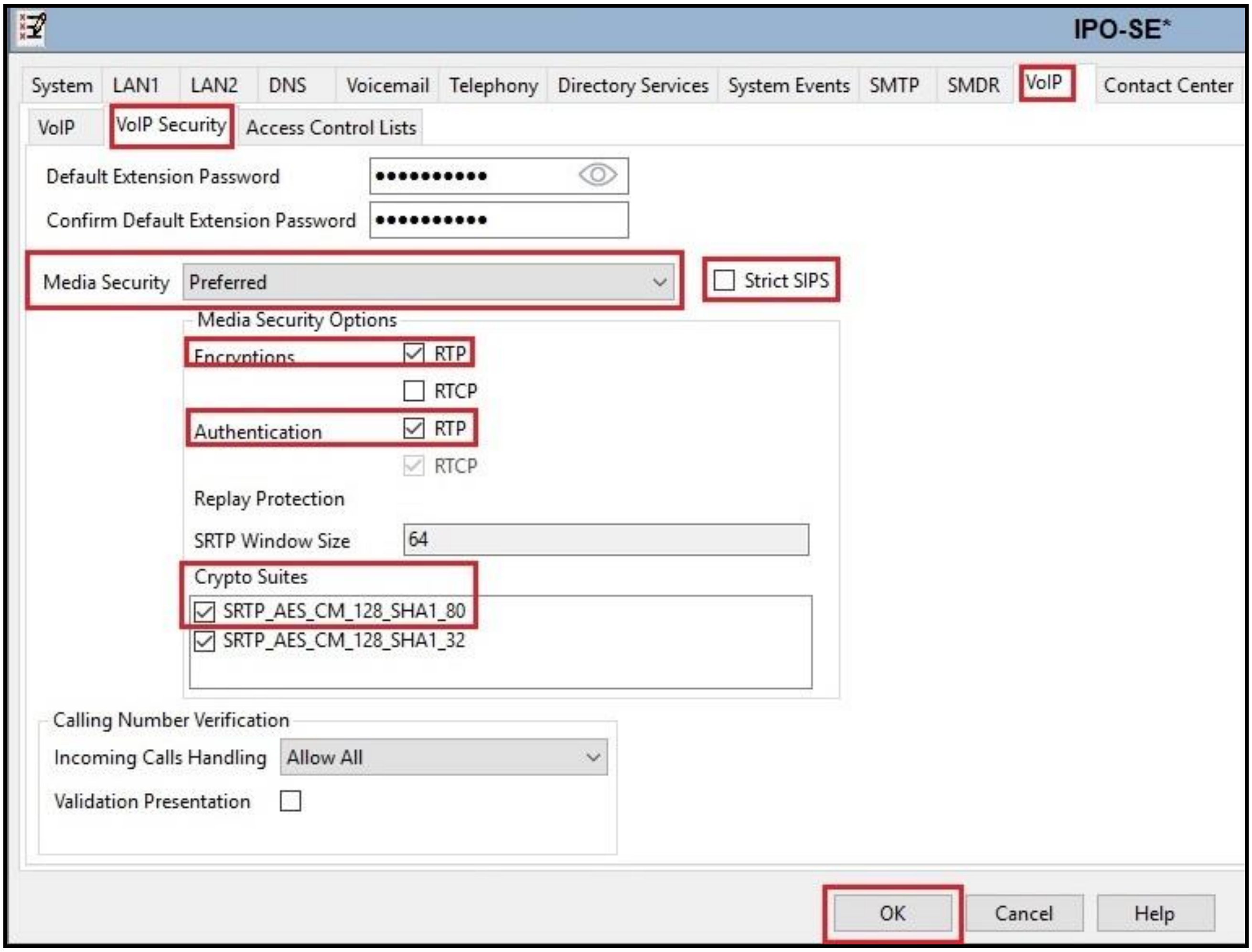


Figure 11 - Avaya IP Office Primary Server VoIP Security

5.4. IP Route

Create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets in order to route calls.

To create an IP route for the Primary system, navigate to **Solution → IPO-SE → IP Route**, right-click on **IP Route** and select **New** (Not shown). The values used during the compliance test are shown below:

- Set the **IP Address and IP Mask** to **0.0.0.0** to make this the default route
- Set **Gateway IP Address** to the IP address of the gateway/router used to route calls to Avaya SBC's network, e.g., **10.33.10.1**
- Set **Destination** to **LAN1** from the pull-down menu
- Click **OK** to commit

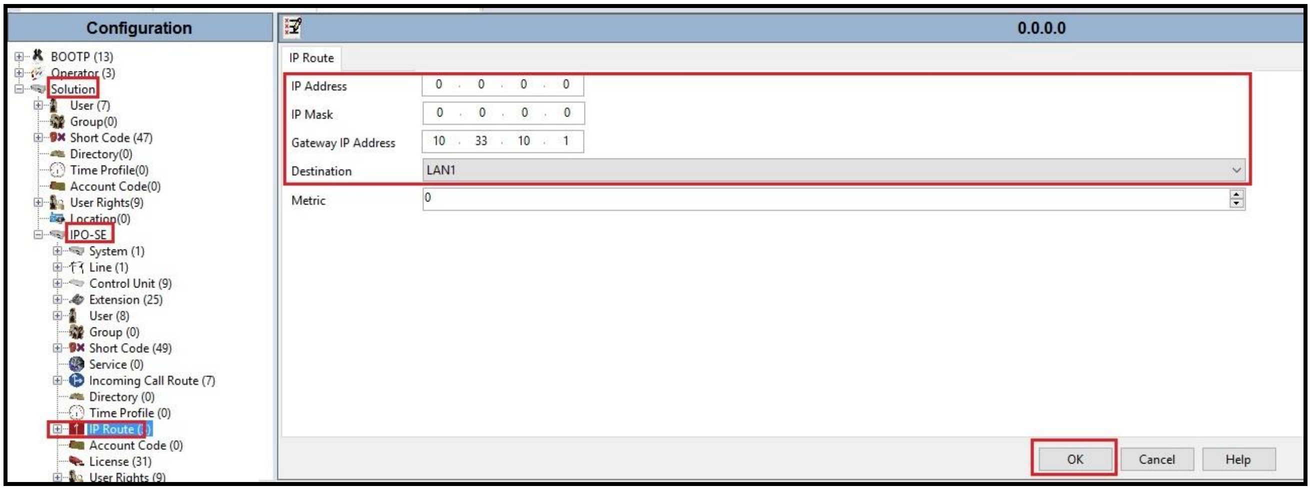


Figure 12 - Avaya IP Office Primary Server IP Route

To create an IP route for the Expansion system, navigate to **Solution → IPOffice_1 → IP Route**, right-click on **IP Route** and select **New** (Not shown). The values used during the compliance test are shown below:

- Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route
- Set **Gateway IP Address** to the IP address of the gateway/router used to route calls to the private network, e.g., **172.16.199.1**
- Set **Destination** to **LAN1** from the pull-down menu
- Click **OK** to commit

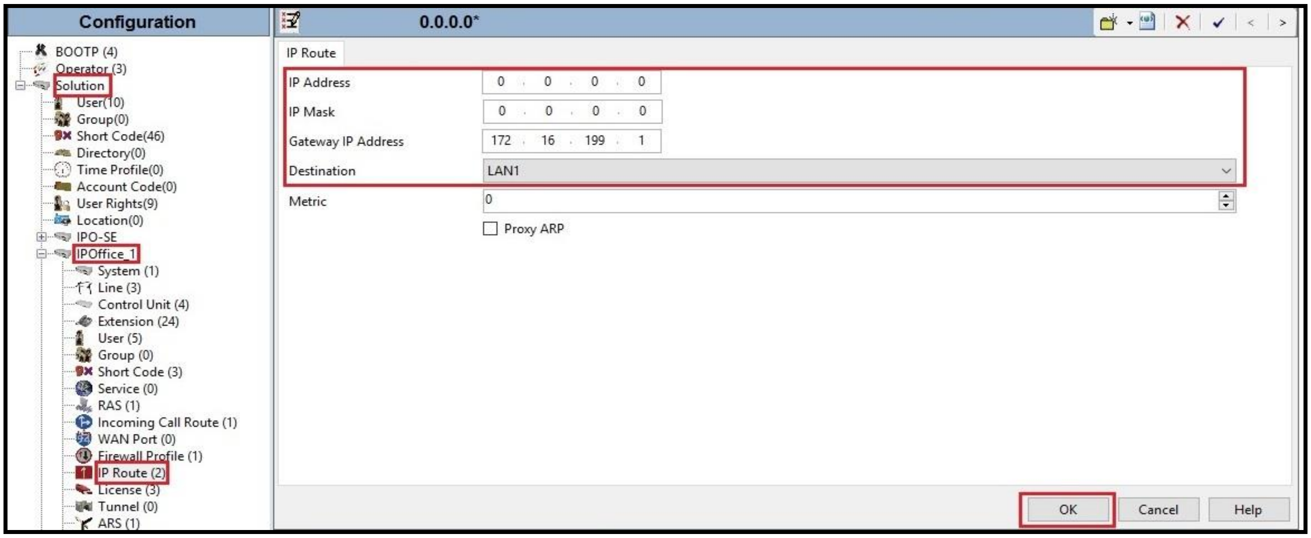


Figure 13 - Avaya IP Office Expansion Server IP Route

5.5. Administer SIP Line

A SIP Line is needed to establish the SIP connection between Avaya IP Office and Cox Communications system via Avaya SBC. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by Avaya IP Office Manager for Server Edition to create a SIP Line. Follow the steps in **Section 5.5.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the Use Network Topology Info field on the Transport tab

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 5.5.2**.

Also, the following SIP Line settings are not supported on Basic Edition:

- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required
- SIP Advanced Engineering

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.5.2**.
For the compliance test, SIP Line 17 was used as trunk for both outgoing and incoming calls.

5.5.1. Create SIP Line from an XML Template

SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment

Create a new folder in a location where Avaya IP Office Server Edition Manager is installed (e.g., C:\Cox Communications\Template). Copy the template file to this folder and rename the template file to **C_IPO12SBC102.xml** (for SIP Line 17).

Create the SIP Trunk from the template, from the Primary server, right-click on **Line** in the Navigation Pane, then navigate to **New from Template → Open from file**.

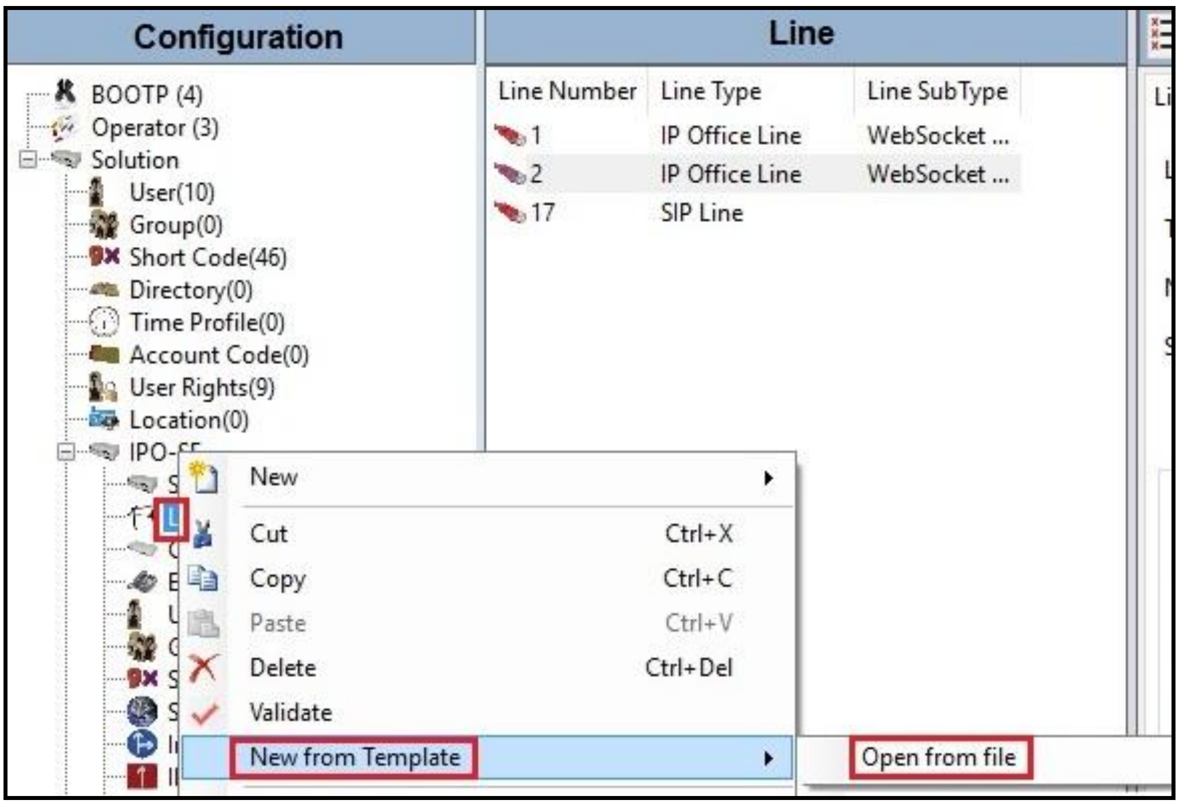


Figure 14 – Create SIP Line from Template

Select the **Template Files (*.xml)** and select the copied template at folder (e.g., C:\Cox Communications\Template). Click **Open** button to create a SIP line from template.

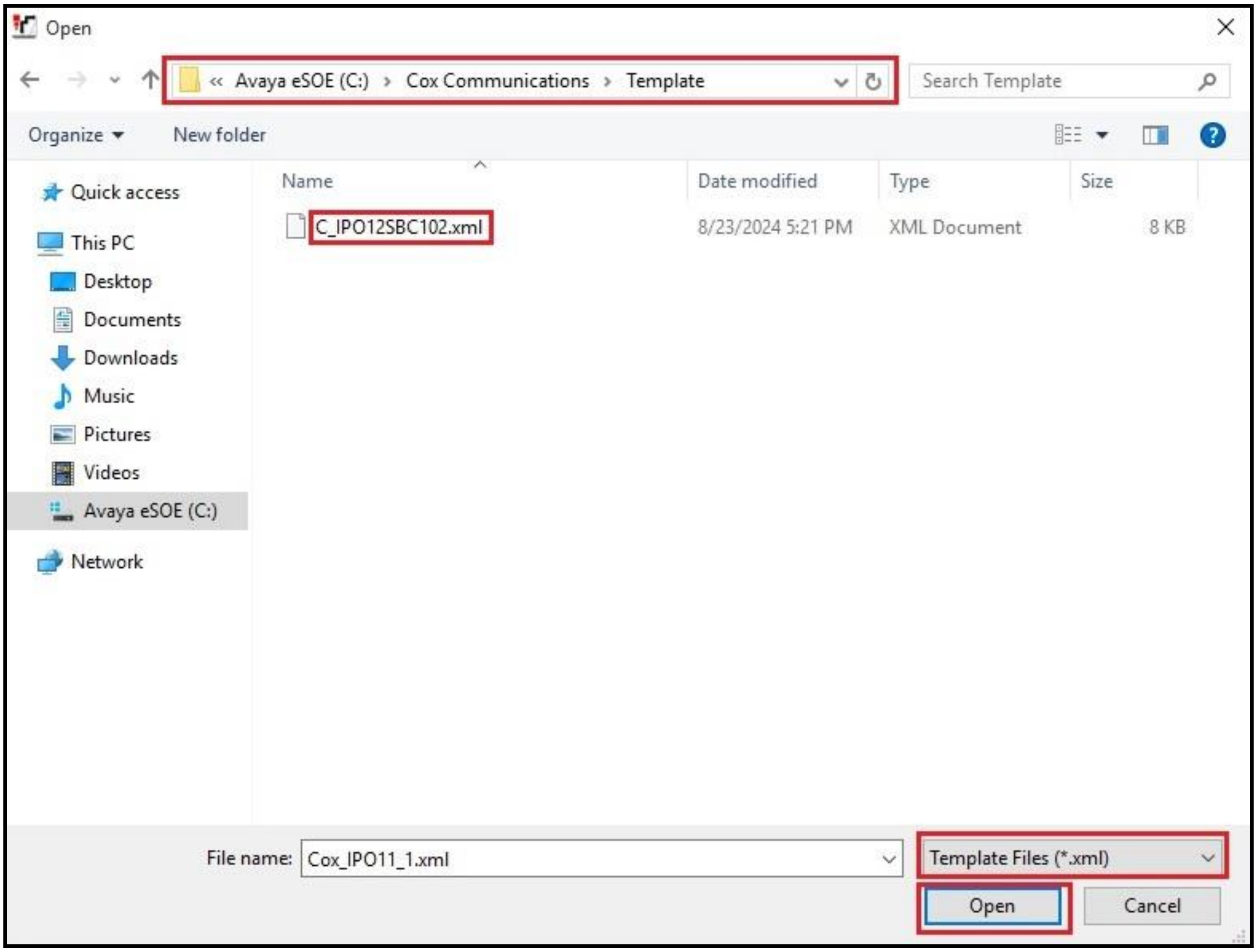


Figure 15 – Create SIP Line from directory

A pop-up window below will appear stating success (or failure). Then click **OK** to continue.

Figure 16 – Create SIP Line from Template successfully

Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.5.2**.

5.5.2. Create SIP Line Manually

To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line** (not shown).

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Select available **Line Number**: **17**
- Check the **In Service** and **Check OOS** box
- Input **ITSP Domain Name**: **10.33.10.49** (This is Avaya SBC internal IP address)
- Input **Local Domain Name**: **10.33.10.56** (This is Avaya IP Office SE LAN1 IP address)
- Set **URI Type** to **SIP URI**
- For **Session Timers**, set **Refresh Method** to **Auto** with **Timer (sec)** to **On Demand**
- Set **Name Priority** to **Favor Trunk**. As described in **Section 5.3.2**, the **Default Name Priority** parameter may retain the default **Favor Trunk** setting or can be configured to **Favor Directory**. As shown below, the default **Favor Trunk** setting was used in the reference configuration
- For **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised Never.** Note: Cox Communications did not support SIP Refer during the compliance testing
- Default values may be used for all other parameters
- Click **OK** to commit then press Ctrl + S to save

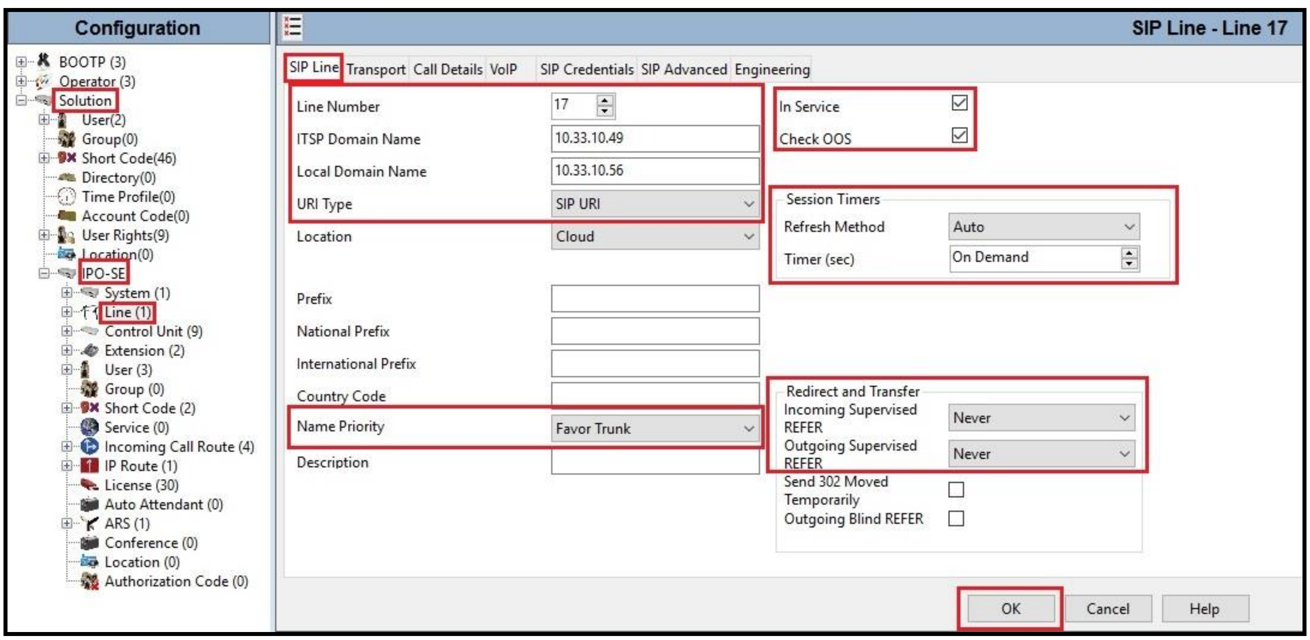


Figure 17 – SIP Line Configuration

On the **Transport** tab in the Details Pane, configure the parameters as shown below:

- The **ITSP Proxy Address** was set to the IP address of Avaya SBC internal interface: **10.33.10.49** as shown in **Figure 1**. This is the SIP Proxy address used for outgoing SIP calls
- In the **Network Configuration** area, **TLS** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5061**
- The **Use Network Topology Info** parameter was set to **None**. The **Listen Port** was set to **5061**. Note: For the compliance testing, the **Use Network Topology Info** field was set to **None**, since no NAT was using in the test configuration. In addition, it was not necessary to configure the **System → LAN1 → Network Topology** tab for the purposes of SIP trunking. If a NAT is used between Avaya IP Office and the other end of the trunk, then the **Use Network Topology Info** field should be set to the LAN interface (**LAN1**) used by the trunk and the **System → LAN1 → Network Topology** tab needs to be configured with the details of the NAT device
- The **Calls Route via Registrar** was unchecked as Cox Communications did not support the dynamic Registration on the SIP Trunk
- Other parameters retain default values
- Click **OK** to commit then press Ctrl + S to save

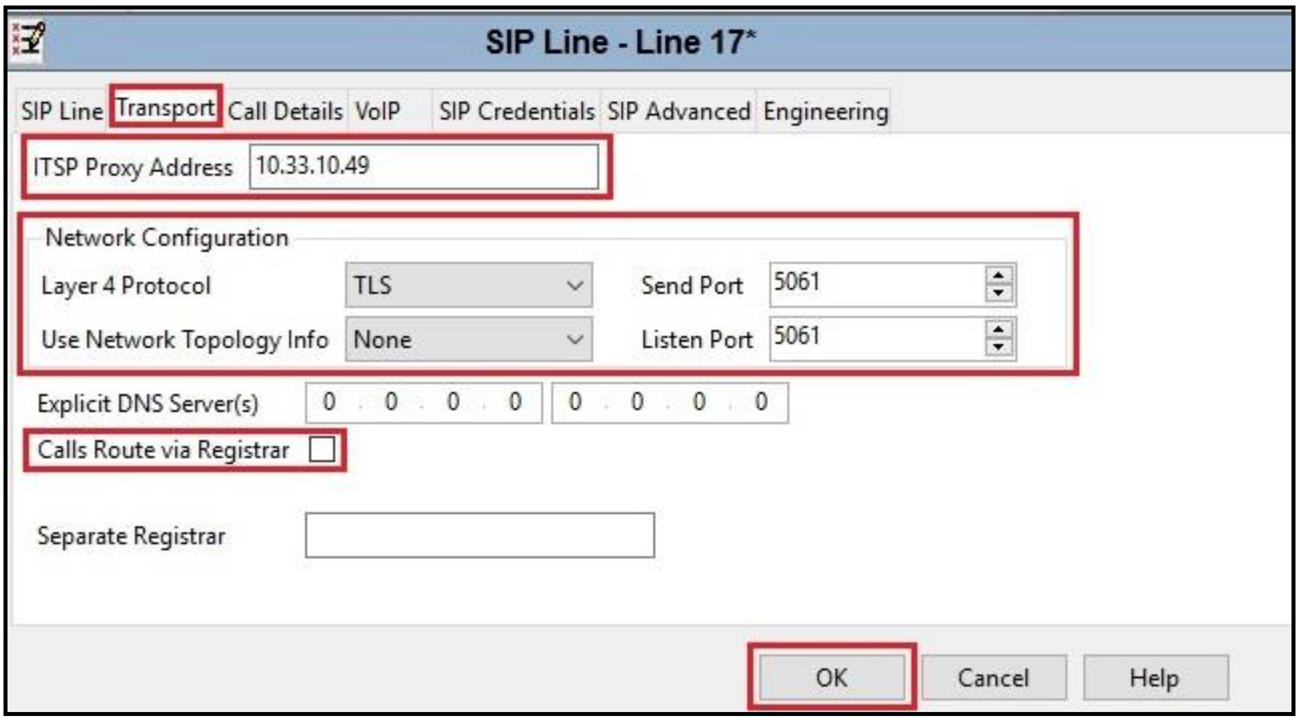


Figure 18 – SIP Line Transport Configuration

The SIP URI entry must be created to match any DID number assigned to an Avaya IP Office user and Avaya IP Office will route the calls on this SIP line. Select the **Call Details** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click **Edit…** button. In the example screen below, a previously configured entry is edited

A SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- Associate this SIP line with an incoming line group in the **Incoming Group** field and an outgoing line group in the **Outgoing Group** field. This line group number will be used in defining incoming and outgoing call routes for this line. For the compliance test, a new line group **17** was defined that only contains this line (line 17)
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern
- Select **Credentials** to **0: <None>**
- Check **P Asserted ID** option
- Check **Diversion Header** option
- Set the **Display** and **Content** of **Local URI**, **Contact**, **P Asserted ID** and **Diversion Header** to **Auto**
- In **Field meaning**: Set **Forwarding/Twinning** of **Local URI**, **Contact**, **P Asserted ID** and **Diversion Header** to **Caller**
- Set all remaining fields as shown on the screenshot below
- Click **OK** to submit the changes

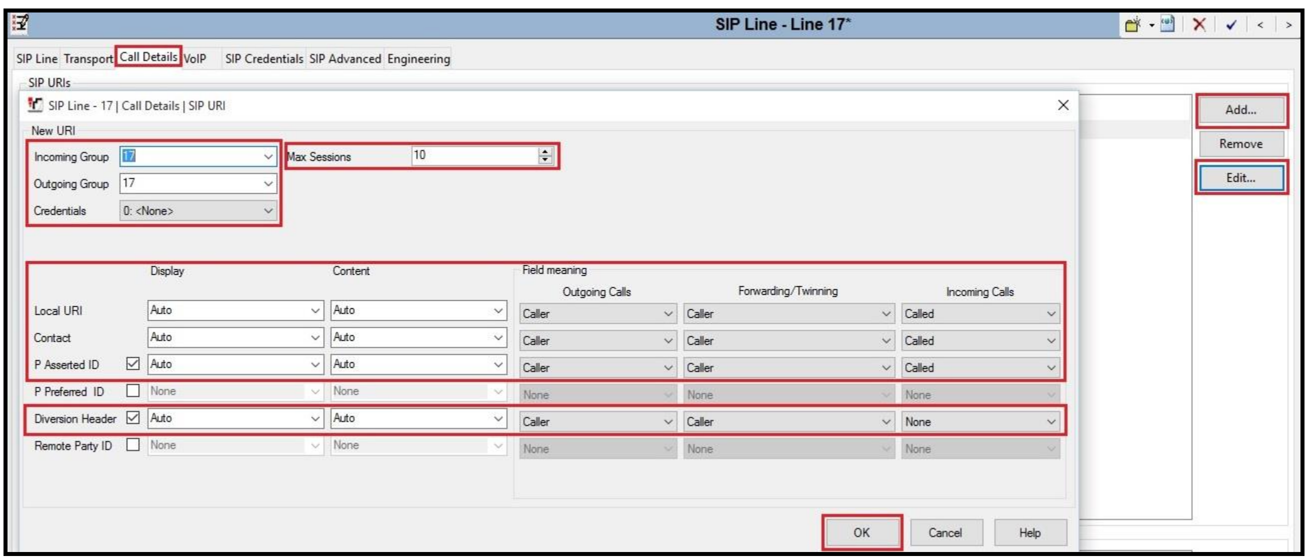


Figure 19 – SIP Line Call Details Configuration

Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. The **G.711 ULAW 64K** codec is selected. Avaya IP Office Server Edition supports the codec, which is sent to the Cox Communications, in the Session Description Protocol (SDP) offer
- Check the **Re-invite Supported** box
- Set **T38 Fallback** from the pull-down menu
- Set the **DTMF Support** to **RFC2833/RFC4733** from the pull-down menu. This directs Avaya IP Office Server Edition to send DTMF tones using RTP events messages as defined in RFC2833 and RFC4733
- Set the **Media Security** field to **Same as System (Preferred)**
- Default values may be used for all other parameters
- Click **OK** to submit the changes

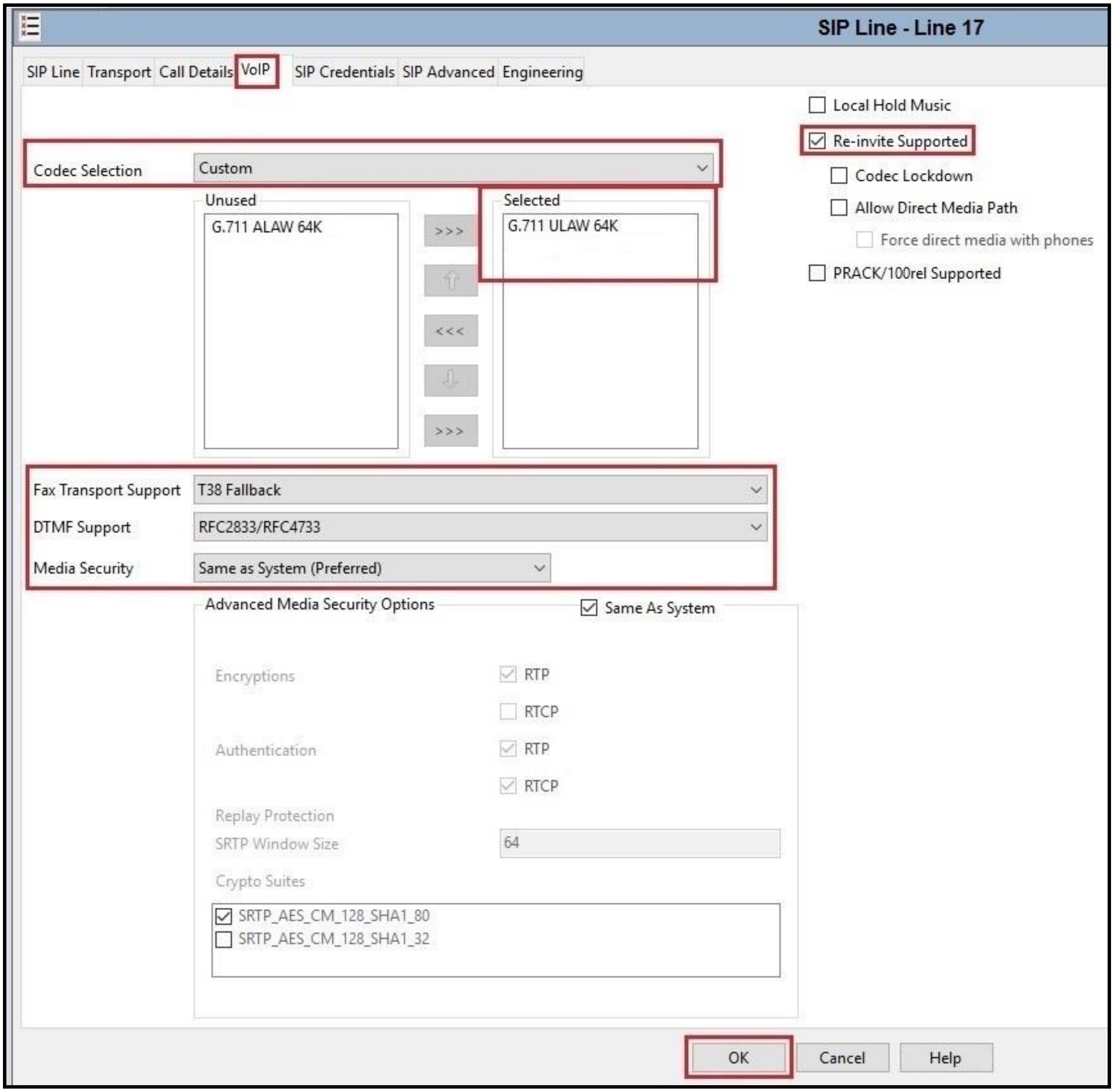


Figure 20 – SIP Line VoIP Configuration

5.6. IP Office Line in Primary System

In IP Office Server Edition systems, IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate line to be configured in the Group pane.

To verify the IP Office line connecting the Primary System to the Expansion System, select **Line** on the navigation pane of Primary System and select the IP Office Line on the Group pane (line **2** on the screen below). Make note of the **Outgoing Group ID 99999** on the Details pane. The **Address** of **Gateway** is Avaya IP Office Expansion System LAN1 IP address **172.16.199.60**.

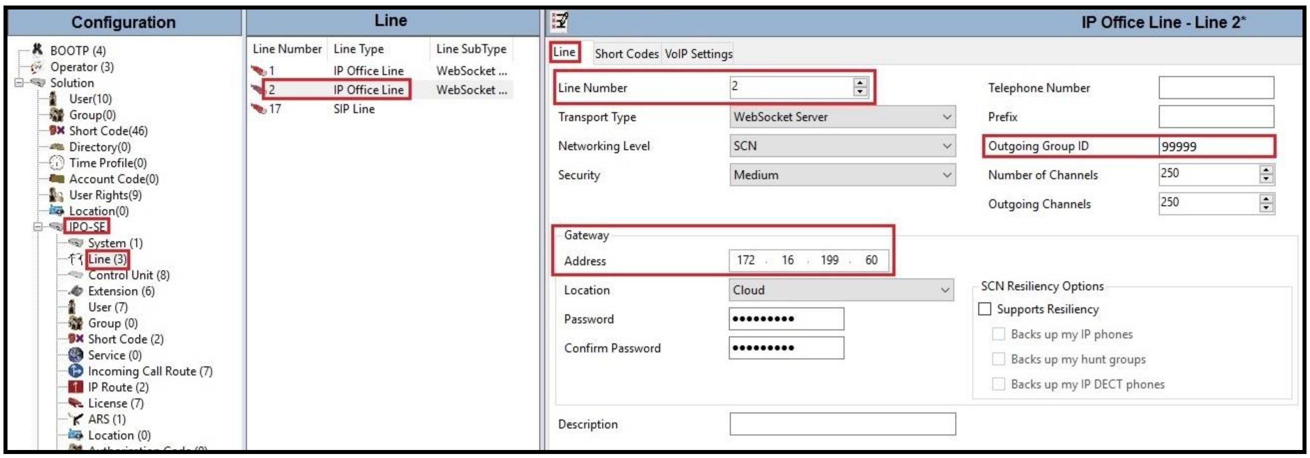


Figure 21 – IP Office Line for Primary System

To verify the **VoIP Settings** of the IP Office line connecting the Primary System to the Expansion System, select **VoIP Settings** tab. The selected codec is **G.711 ULAW 64K**. Select **Fax Transport Support** to **T38 Fallback** (This setting should be as same as the VoIP settings in SIP line of Primary System and the VoIP settings in IP Office Line of Expansion System). Default values may be used for all other parameters. Click **OK** to submit the changes.

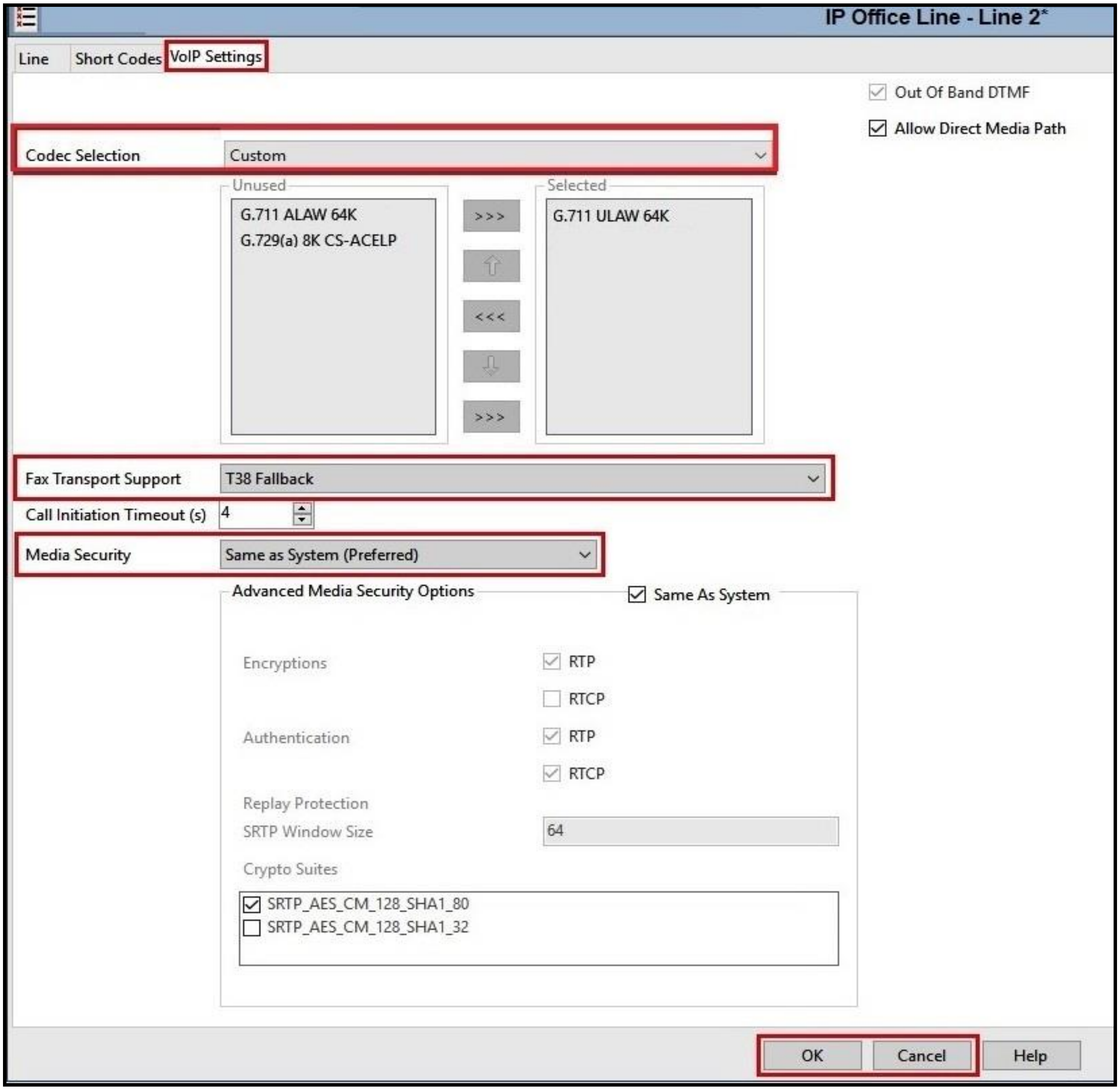


Figure 22 – IP Office Line for Primary System VoIP Settings

5.7. IP Office Line in Expansion System

To verify the IP Office line connecting the Expansion System to the Primary System, select Expansion Line on the navigation pane and select the IP Office Line on the Group pane (line **17** on the screen below). Make note of the **Outgoing Group ID 99999** on the Details pane. The **Address** of **Gateway** is Avaya IP Office Server Edition LAN1 IP address **10.33.10.56**.

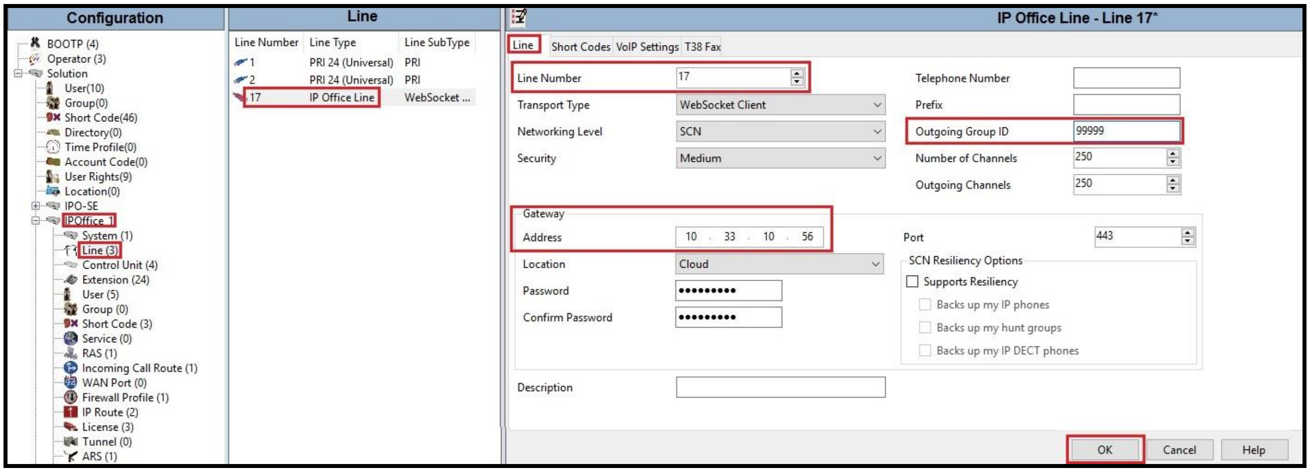


Figure 23 – IP Office Line for Expansion System

To verify the **VoIP Settings** of the IP Office line connecting the Expansion System to the Primary Server, select **VoIP Settings** tab. The selected codec is **G.711 ULAW 64K**. Select **Fax Transport Support** to **T38 Fallback** (This setting should be as same as the VoIP settings in SIP line and IP Office Line of Primary System). Default values may be used for all other parameters. Click **OK** to submit the changes.

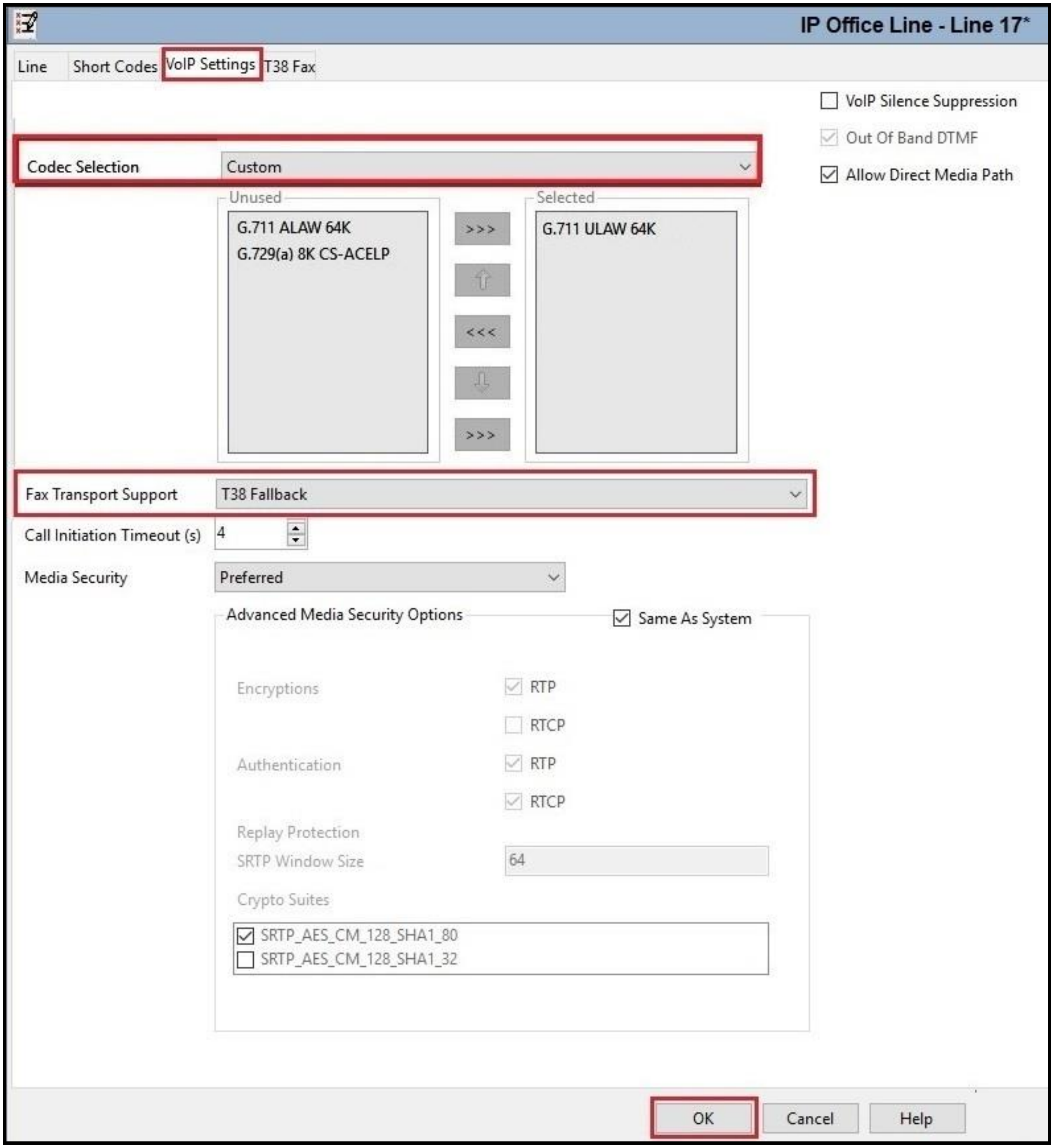


Figure 24 – IP Office Line for Expansion Server VoIP Settings

To verify the **T38 Fax** of the IP Office line connecting the Expansion System to the Primary Server, select **T38 Fax** tab (Note: The T38 Fax tab is only active when Fax Transport Support is selected as T38 Fallback on VoIP Settings tab). Uncheck the **Use Default Values** at the bottom of the screen. Set the **T.38 Fax Version** to **0**. Default values may be used for all other parameters. Click the **OK** to submit the changes.

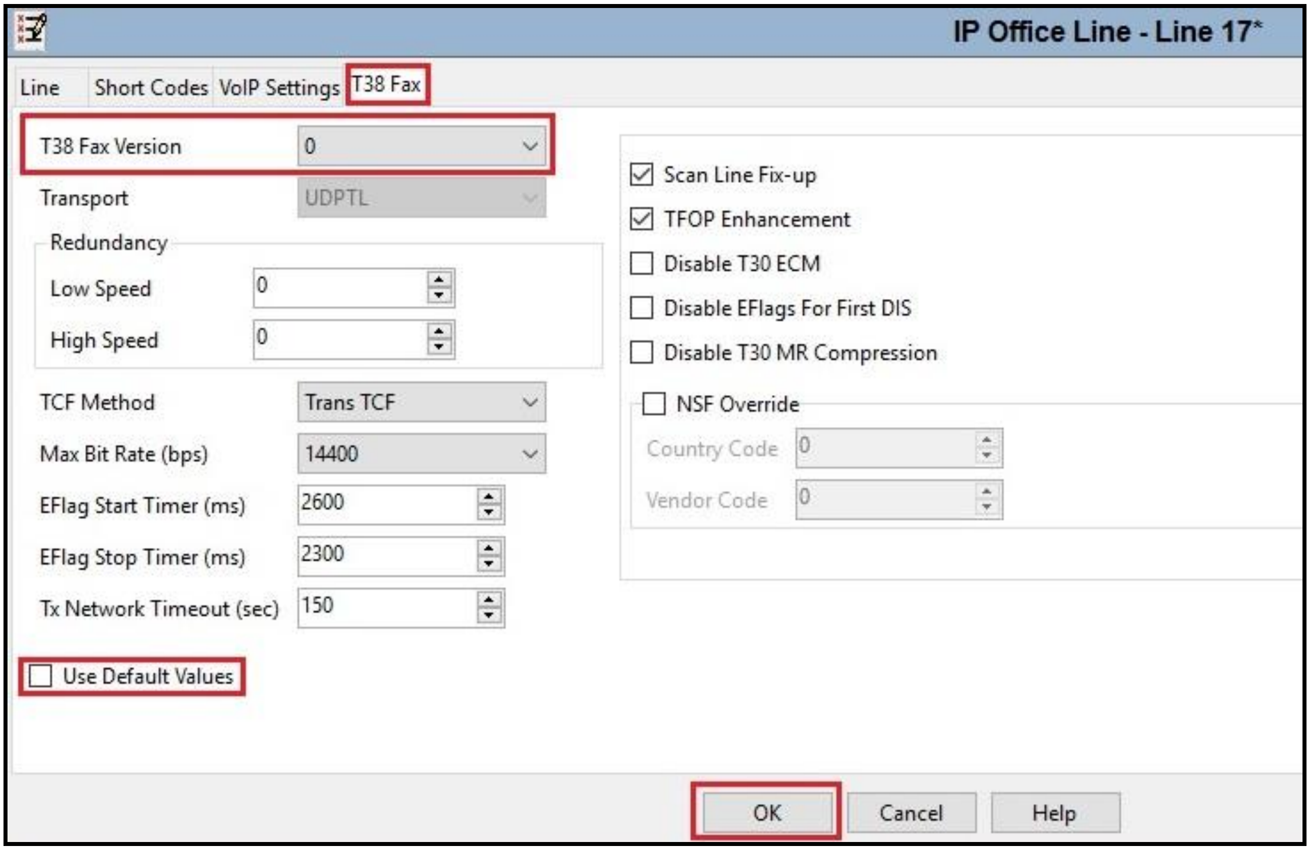


Figure 25 – IP Office Line for Expansion Server T38 Fax

5.8. Outbound Short Code

Define a short code to route outbound traffic on the SIP line to Cox Communications. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created.

The screen below shows the details of the previously administered "**9N;**" short code for Primary System used in the test configuration.
Navigate to **Solution → IPO-SE → Short Code**, right-click on **Short Code** and select **New**.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**, this short code will be invoked when the user dials 9 followed by any number

- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to **N**. The value **N** represents the number dialed by the user. Note: Use the specific **W** in front of **N** for restricting all outbound calls
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.5.2**. This short code will use this line group when placing the outbound call
- Set the **Locale** to **United State (US English)**
- Default values may be used for all other parameters
- Click **OK** to submit the changes

Figure 26 – Short Code 9N for Primary System

The screen below shows the details of the previously administered "**9N;**" short code for Expansion System used in the test configuration.

Navigate to **Solution → IPOffice_1 → Short Code**, right-click on **Short Code** and select **New**

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**, this short code will be invoked when the user (using Avaya analog or digital phones) dials 9 followed by any number
- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to **9N**
- Set the **Line Group ID** to **99999** defined on the **Outgoing Group ID** of the IP Office line connecting the Expansion System to the Primary System. This short code will use this line group when placing the outbound call via Avaya IP Office Server Edition Primary Server
- Default values may be used for all other parameters
- Click **OK** to submit the changes

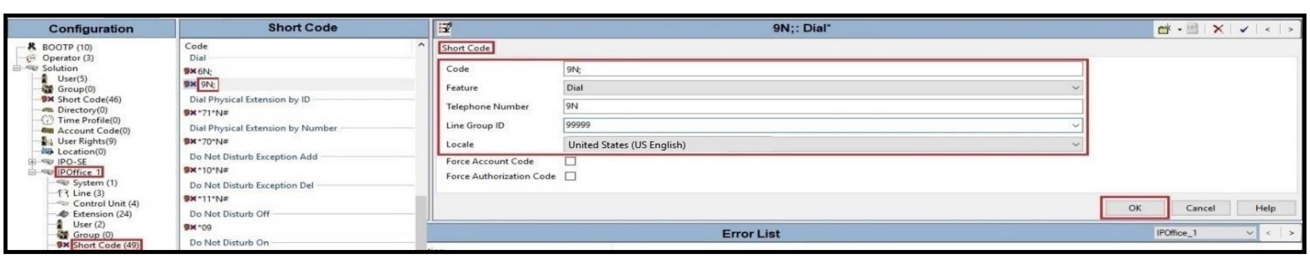


Figure 27 – Short Code 9N for Expansion System

The feature of incoming calls from mobility extension to idle-appearance FNE (Feature Name Extension) is hosted by Avaya IP Office Server Edition. The screen below shows the details of the previously administered "**77700**" short code for FNE Service.
Navigate to **Solution → Short Code**, right-click on **Short Code** and select **New**. The feature of incoming calls from mobility extension to idle-appearance FNE (Feature Name Extension) is hosted by Avaya IP Office Server Edition. The Short Code **77700** was configured with following parameters:

- For **Code** field, enter FNE feature code as **77700** for dial tone
- Set **Feature** to **FNE Service**
- Set **Telephone Number** to **00**
- Set **Line Group ID** to **0**
- Set the **Locale** to **United States (US English)**
- Default values may be used for other parameters
- Click **OK** to submit the changes

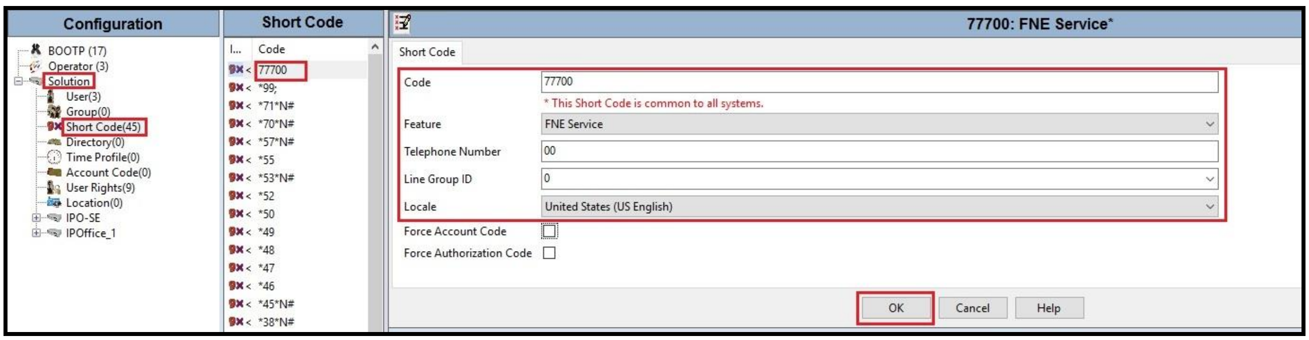


Figure 28 – Short Code FNE 77700

5.9. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line defined in **Section 5.5.2**. To configure these settings, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **531XXX4176**. Select the **User** tab in the Details pane.

Note: When **Auto** is selected for the **Local URI**, **Contact** and **Diversion Header** parameters (See **Section 5.5.2 - Call Detail** tab), the information in the Incoming Call Route (See **Section 5.10**) is used to populate the SIP From and Contact headers for outbound calls.

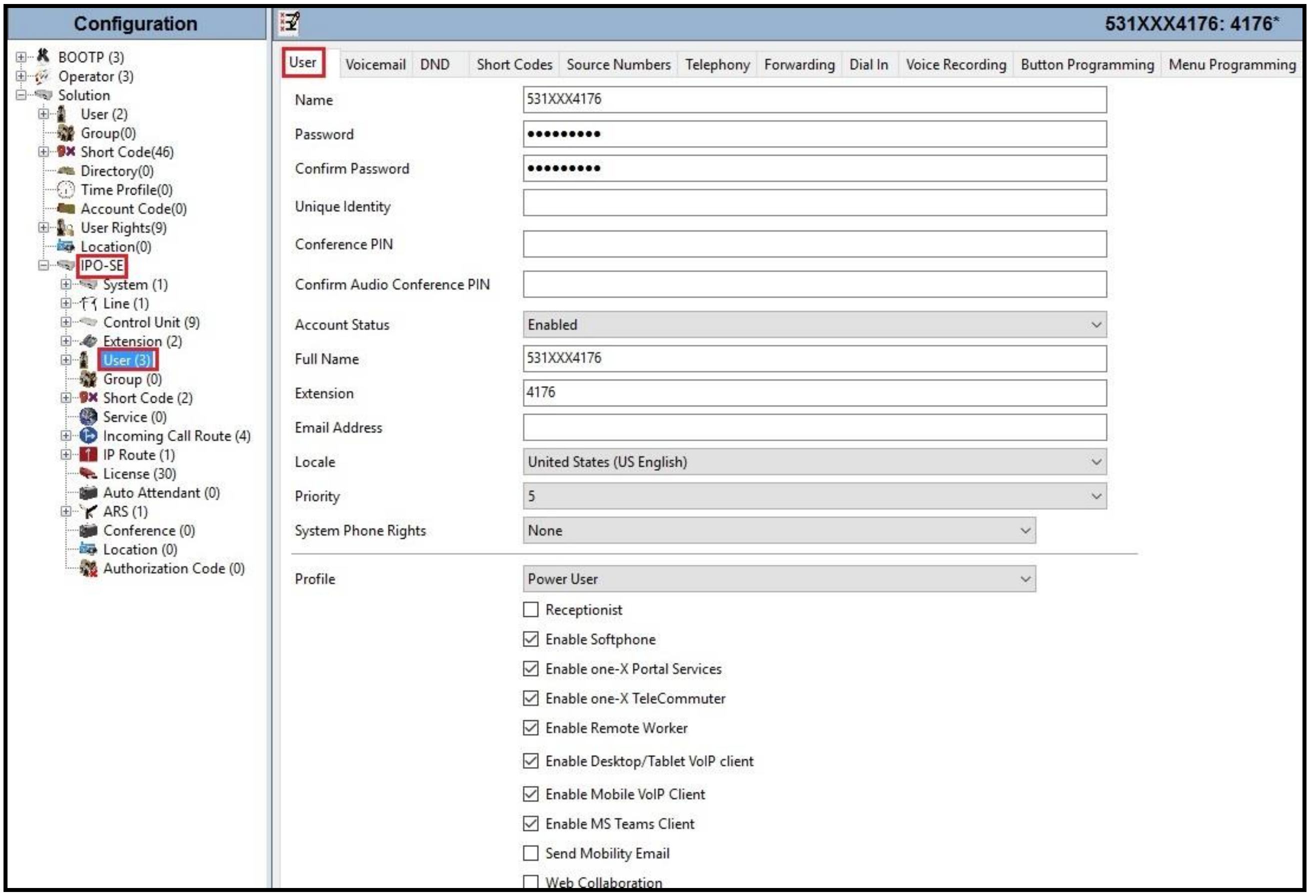


Figure 29 – User Configuration – User tab

To configure the restricted outbound call for a user by using specific W in the Short Code, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **531XXX4176**. Select the **Short Codes** tab in the Details pane.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**, this short code will be invoked when the user dials 9 followed by any number
- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to **WN**. The value **N** represents the number dialed by the user. Note: Use the specific **W** in front of **N** for restricting outbound calls for a user
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.5.2**. This short code will use this line group when placing the outbound call
- Set the **Locale** to **United State (US English)**
- Default values may be used for all other parameters
- Click **OK** to submit the changes

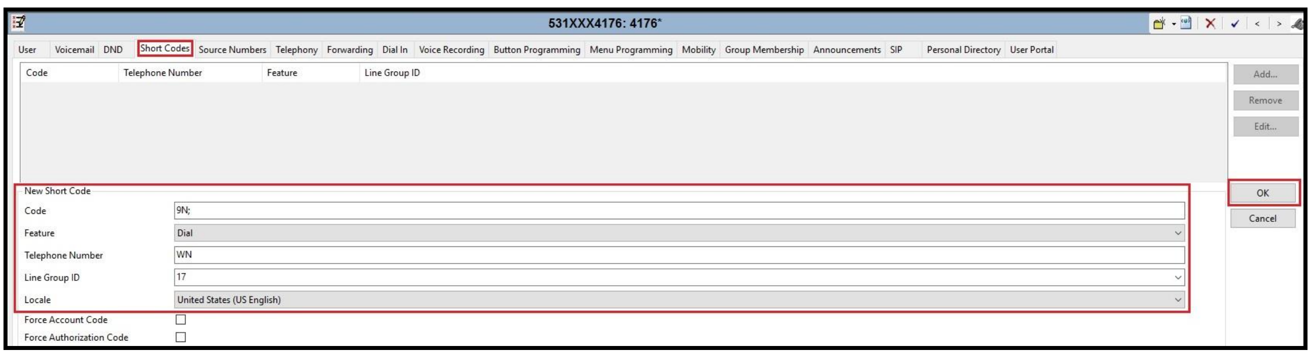


Figure 30 – User Configuration – Short Code tab

One of the H.323 IP Deskphones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for **User 531XXX4176**. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **91613XXX5096**. Check **Mobile Call Control** to allow incoming calls from mobility extension to access 77700 (Defined in **Section 5.8**). Other options can be set according to customer requirements.

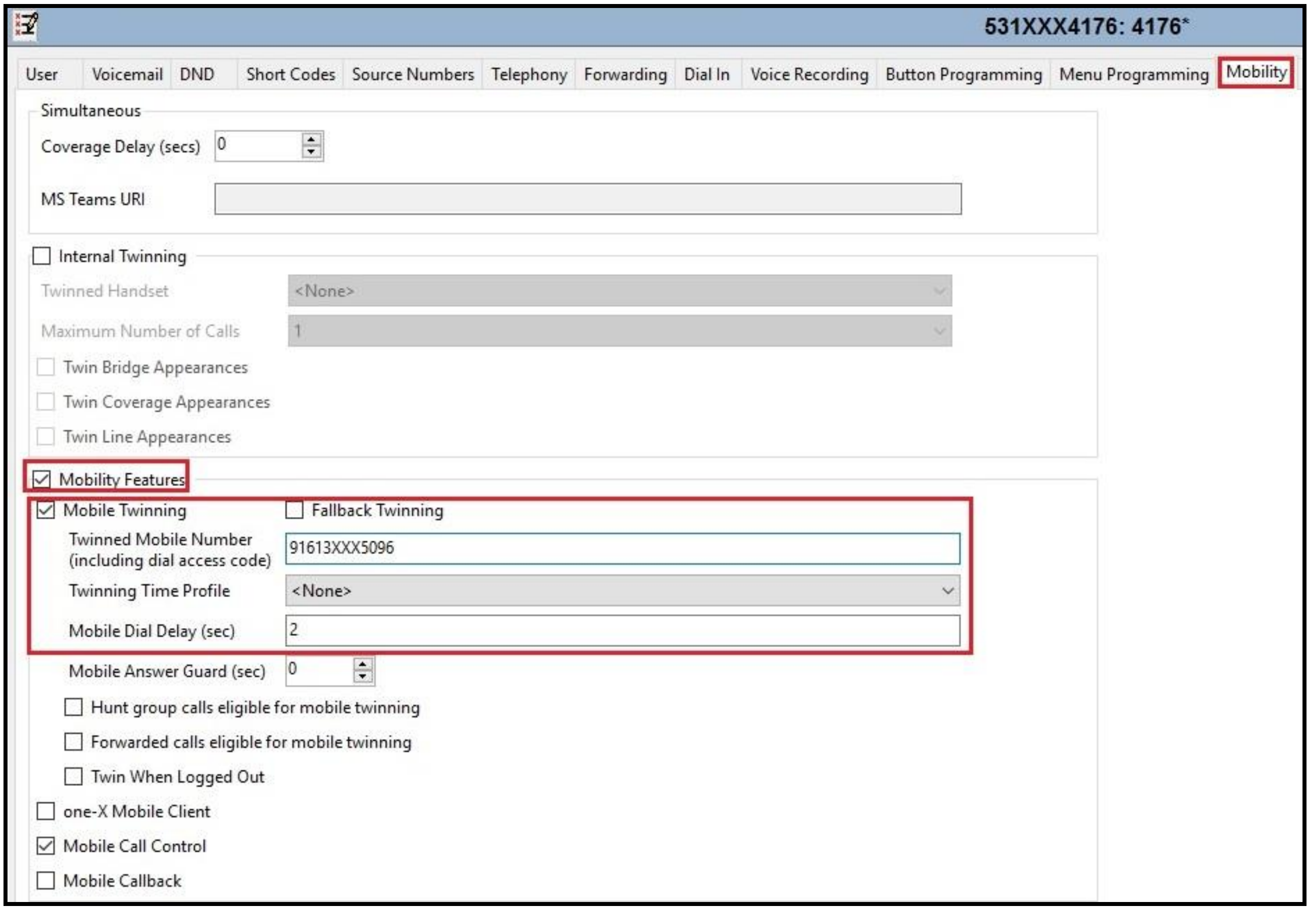


Figure 31 – Mobility Configuration for User

5.10. Incoming Call Route

An Incoming Call Route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New** (not shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capability** to **Any Voice**
- Set the **Line Group ID** to the **Incoming Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.5.2**
- Set the **Incoming Number** to the incoming DID number on which this route should match
- Default values can be used for all other fields

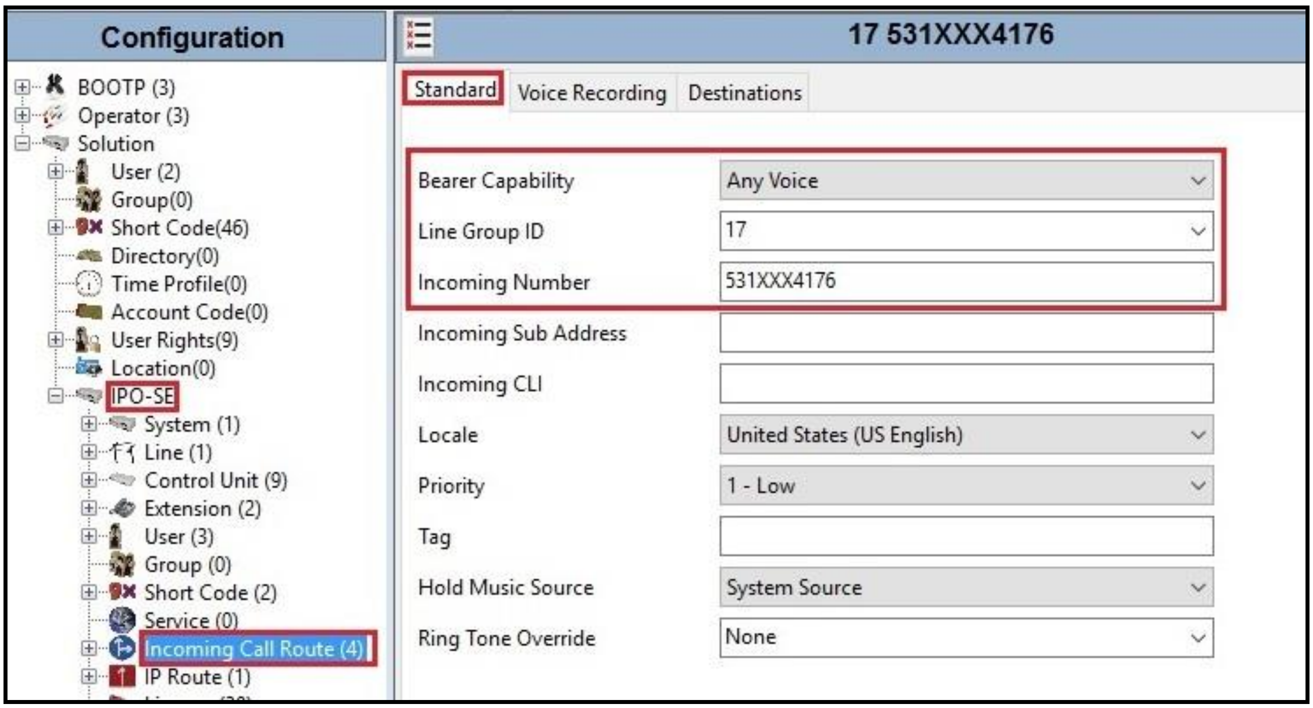


Figure 32 – Incoming Call Route Configuration

On the **Destination** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **531XXX4176** on line 17 are routed to **Destination 4176 531XXX4176** as below screenshot:

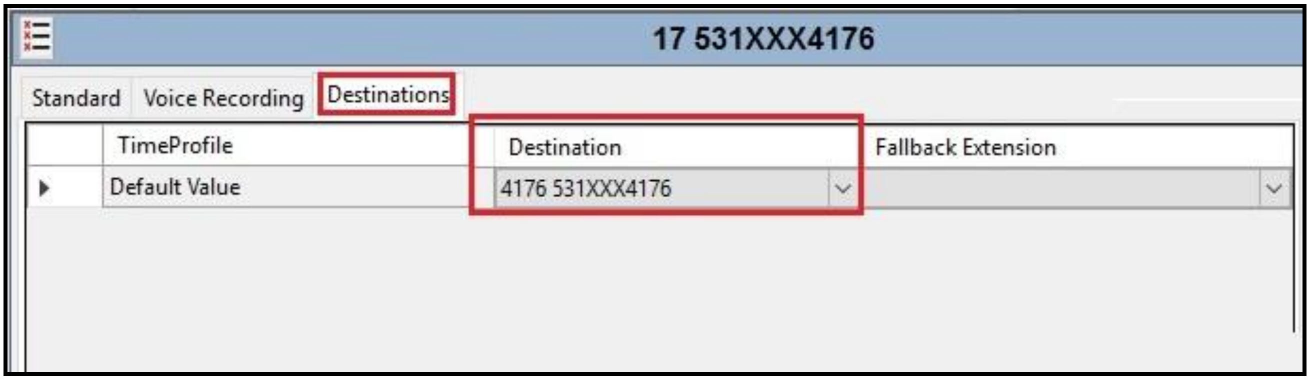


Figure 33 – Incoming Call Route for Destination 531XXX4176

For Feature Name Extension Service testing purpose, the incoming calls to DID number **531XXX7634** were configured to access to FNE Service. The **Destination** was appropriately defined as **77700** as below screenshot:

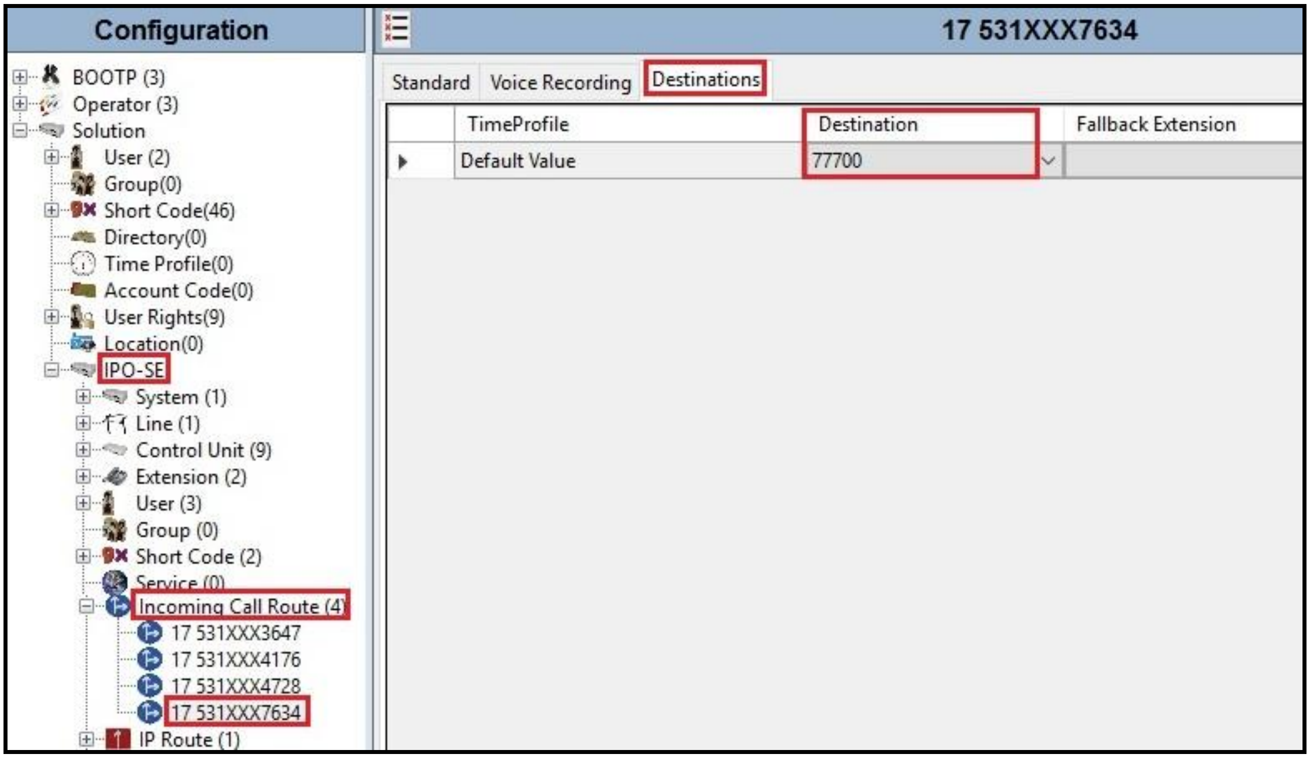


Figure 34 – Incoming Call Route for Destination FNE

For Voice Mail testing purpose, the incoming calls to DID number **531XXX3647** were configured to access **VoiceMail**. The **Destination** was appropriately defined as **VoiceMail** as below screenshot:

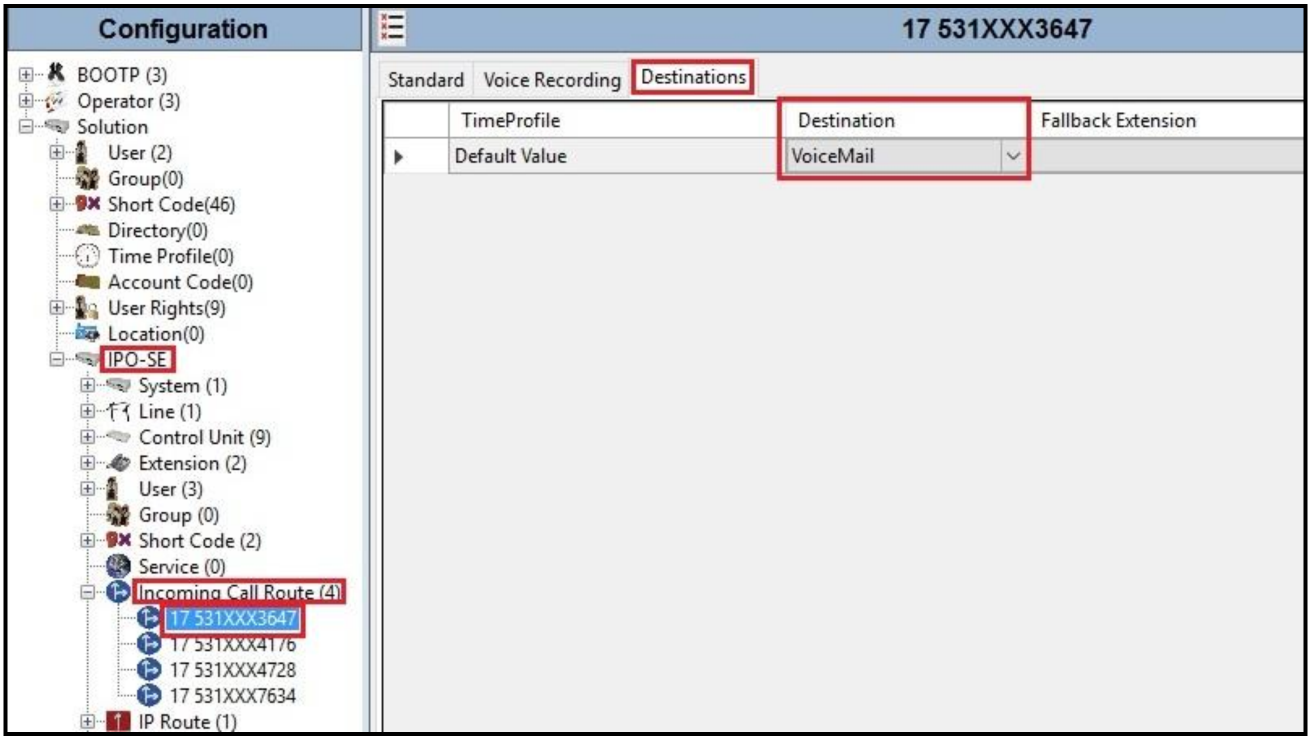


Figure 35 – Incoming Call Route for Destination VoiceMail

5.11. Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

6. Configure Avaya Session Border Controller

This section describes the configuration of Avaya SBC necessary for interoperability with the Avaya IP Office and Cox Communications SIP Trunk Service.

Avaya elements reside on the Private side and the Cox Communications SIP Trunk Service resides on the Public side of the network, as illustrated in **Figure 1**.

Note: The following section assumes that Avaya SBC has been installed and that network connectivity exists between the systems. For more information on Avaya SBC, see relevant product documentation references in **Section 10** of these Application Notes.

6.1. Log in to the Avaya SBC

Access the web interface by typing "**https://x.x.x.x/sbc/**" (where x.x.x.x is the management IP address of the Avaya SBC).

Enter the **Username** and **Password**

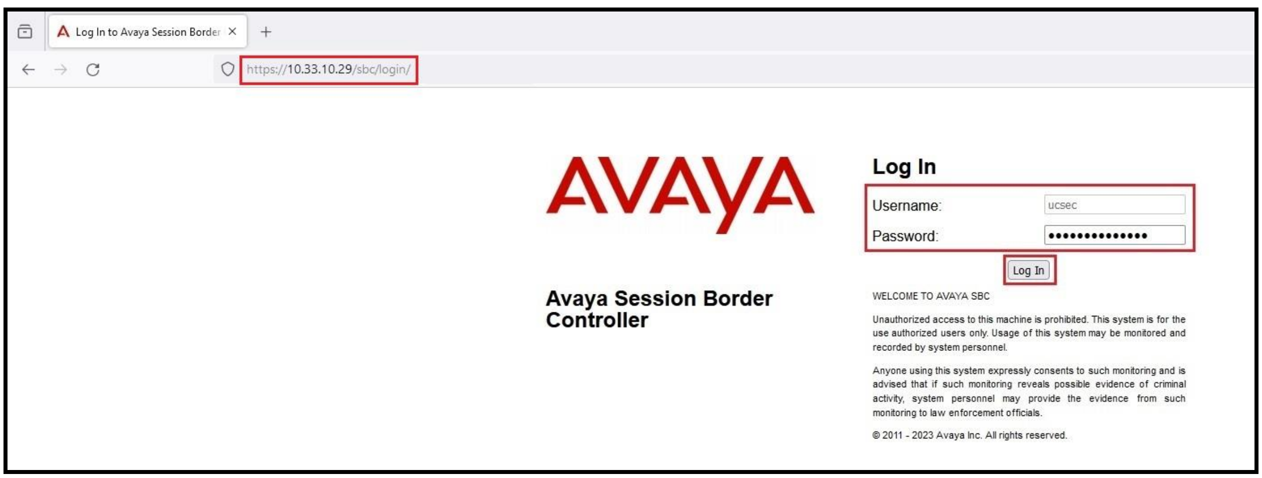


Figure 36 – Avaya SBC Login

The **Dashboard** main page will appear as shown below.

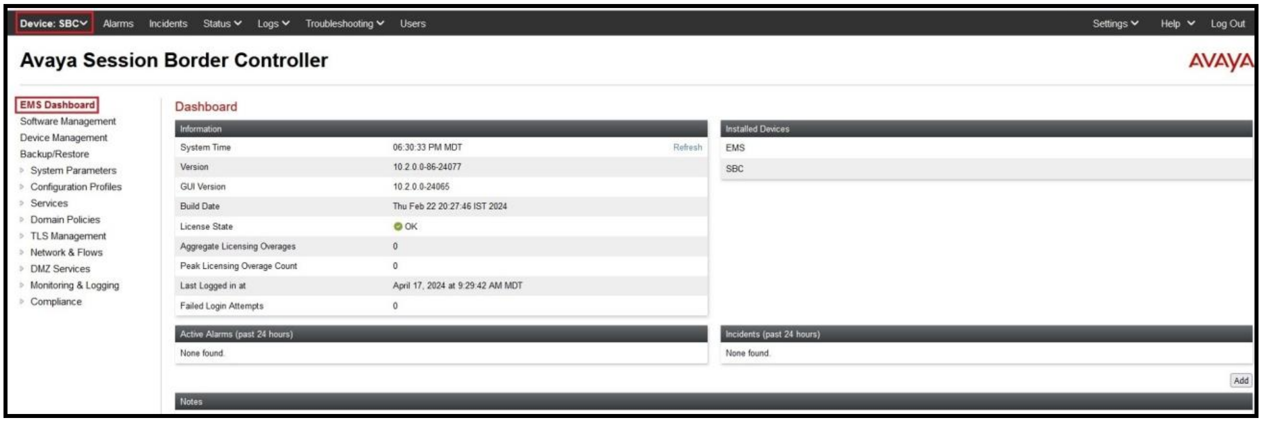


Figure 37 - Avaya SBC Dashboard

To view system information that has been configured during installation, navigate to **Device Management**. A list of installed devices is shown in the right pane. In the compliance test, a single Device Name **SBC** was already added. To view the configuration of this device, click **View** as shown in the screenshot below.

Figure 38 - Avaya SBC Device Management

The **System Information** screen shows **General Configuration**, **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**.

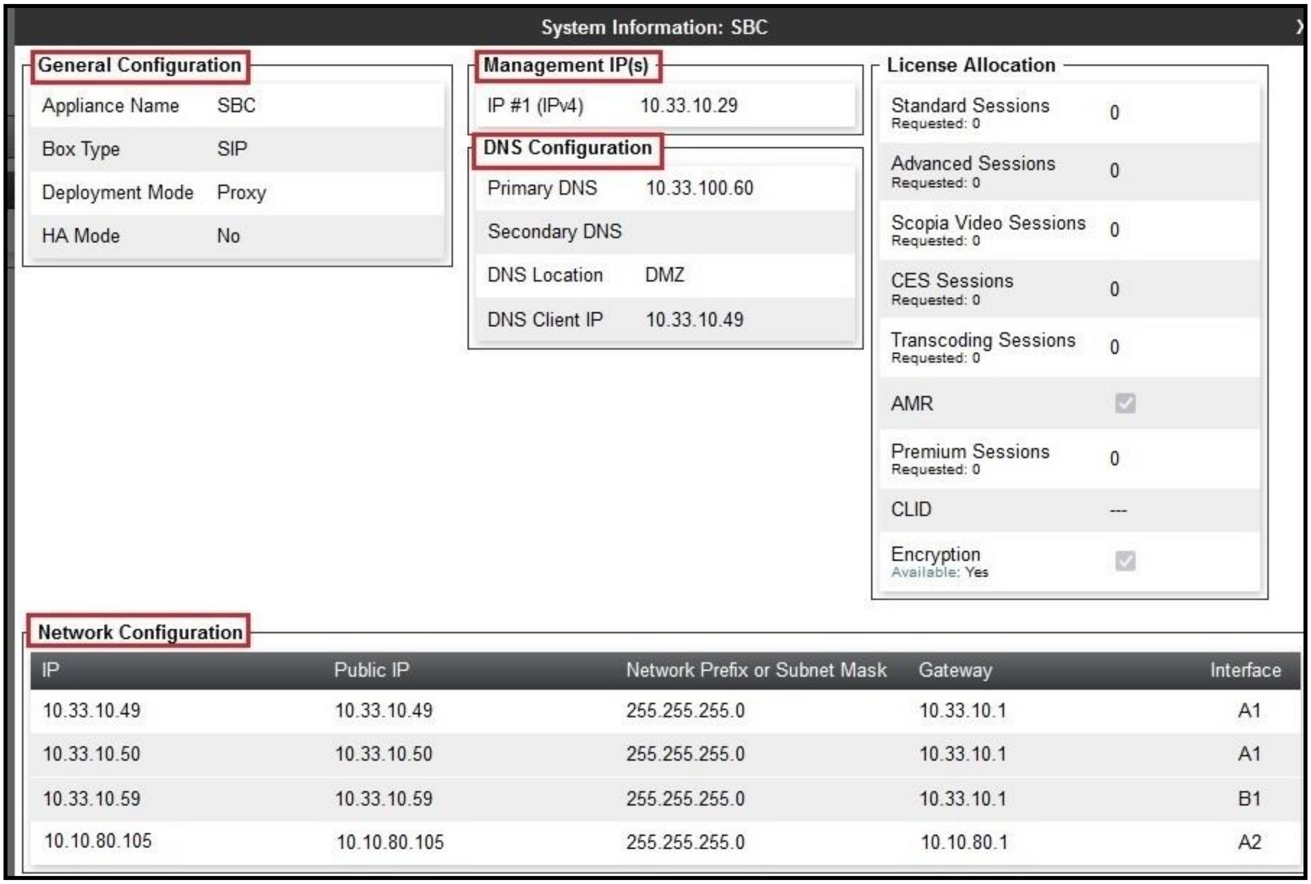


Figure 39 - Avaya SBC System Information

6.2. Server Interworking

Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server).

6.2.1. Configure Server Interworking Profile – Avaya IP Office

Server Interworking profile allows administrator to configure and manage various SIP call server-specific capabilities such as call hold, 180 handling, etc.

From the menu on the left-hand side, select **Configuration Profiles → Server Interworking**

- Select **avaya-ru** in **Interworking Profiles**
- Click **Clone**
- Enter **Clone Name**: **IPO** and click **Finish** (not shown)
- Click **Edit** button
- Check **T.38 Support** option and click **Finish** (not shown).

The following screen shows that Avaya IP Office server interworking profile (named: **IPO**) was added.

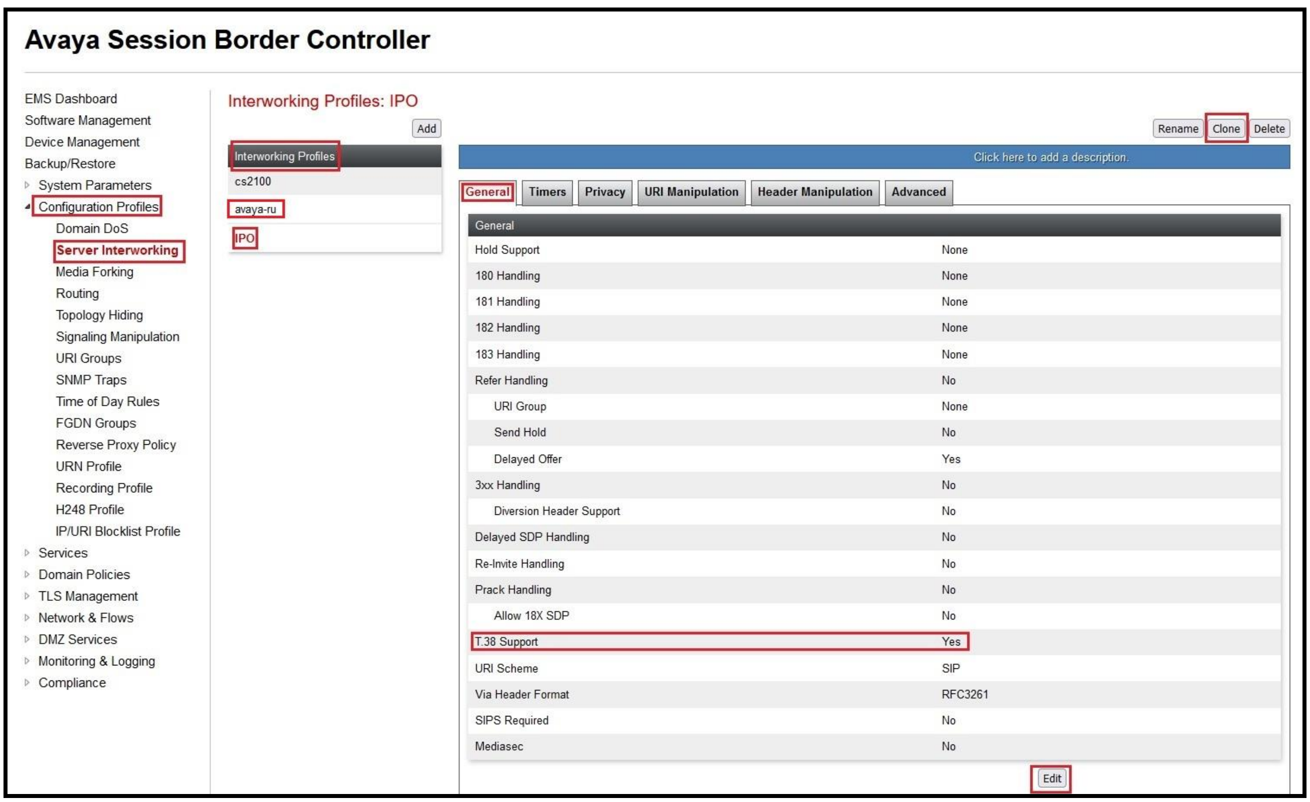


Figure 40 - Server Interworking – Avaya

6.2.2. Configure Server Interworking Profile – Cox Communications

From the menu on the left-hand side, select **Configuration Profiles → Server Interworking → Add**

- Enter **Profile Name**: **SP** (not shown)
- Click **Next** button to leave all options at default and click **Finish** (not shown)
- Click **Edit** button
- Check **T.38 Support** option and click **Finish** (not shown)

The following screen shows that Cox Communications server interworking profile (named: **SP**) was added.

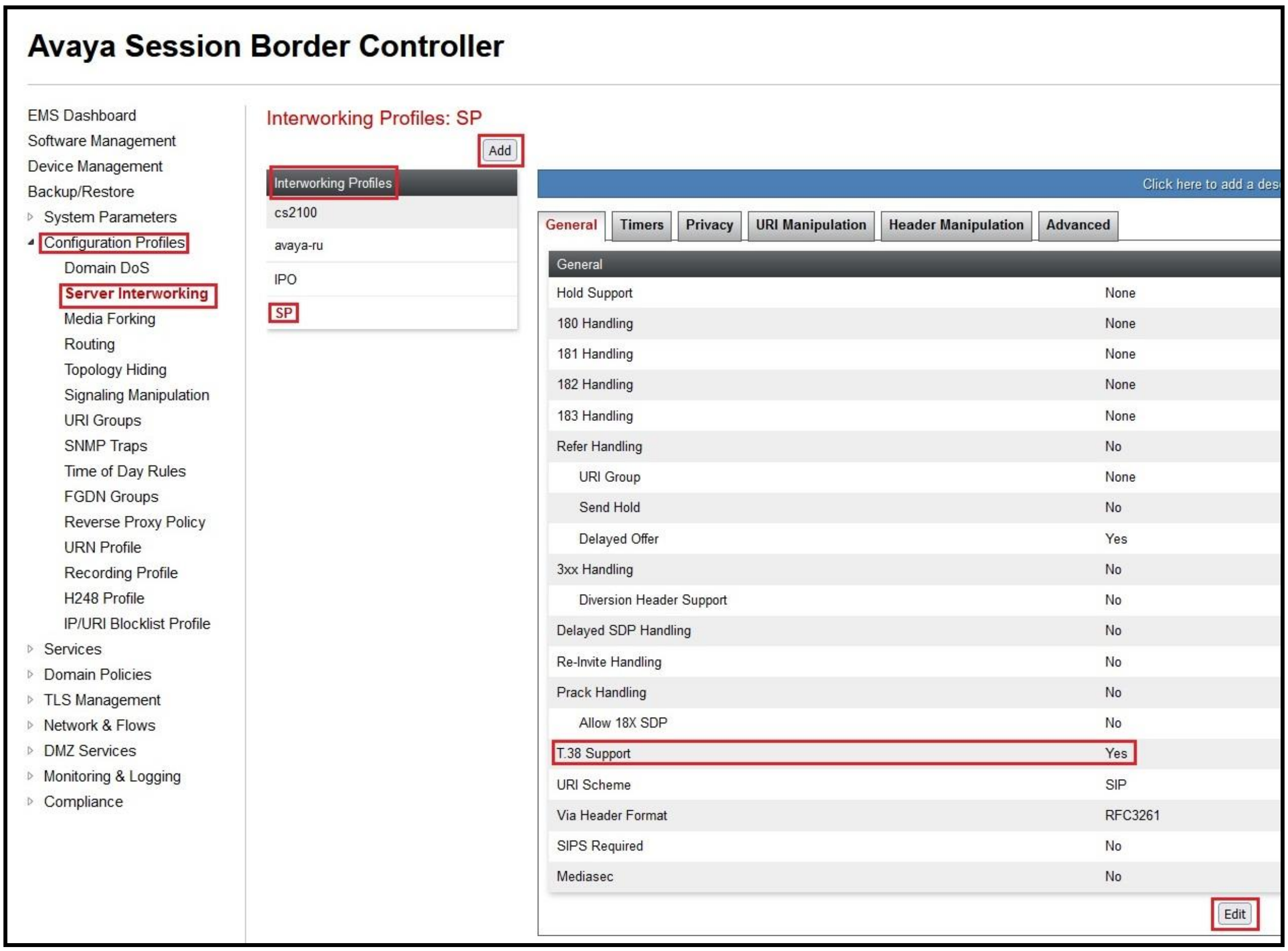


Figure 41 - Server Interworking – Cox Communications

6.3. Configure SIP Server

Servers are defined for each server connected to the Avaya SBC. In this case, IP Office is connected as the Call Server and Cox Communications is connected as the Trunk Server

6.3.1. Configure SIP Server – Avaya Site

The **SIP Servers** screen contains six tabs: **General**, **Authentication**, **Heartbeat**, **Registration**, **Ping** and **Advanced**. Together, these tabs allow one to configure and manage various SIP call server specific parameters such as port assignment, IP Server type, heartbeat signaling parameters and some advanced options

From the menu on the left-hand side, select **Services → SIP Servers → Add**

Enter **Profile Name**: **IPO**

On **General** tab, enter the following:

- **Server Type**: Select **Call Server**
- **TLS Client Profile**: Select **AvayaSBCClient**. Note: During the compliance test in the lab environment, demo certificates are used on Avaya Aura Session Manager, and are not recommended for production use.
- **IP Address/FQDN**: **10.33.10.56** (IP Office SE LAN1 IP address)
- **Port**: **5061**
- **Transport**: **TLS**
- Click **Finish** (not shown)

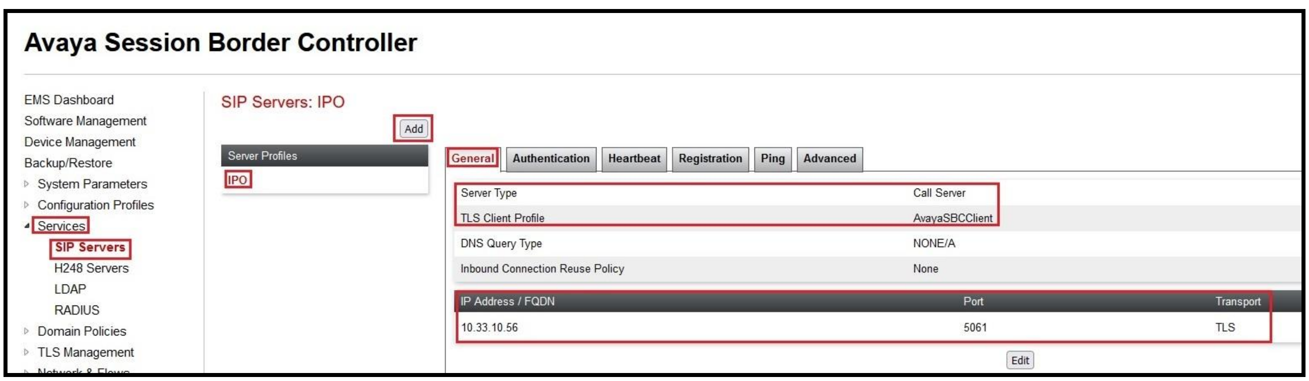


Figure 42 – Avaya SIP Server Configuration – General

On the **Heartbeat** tab, click **Edit** button to enter the following:

- Check **Enable Heartbeat** option
- **Method**: **OPTIONS**
- Set **Retry Timeout on Connection Failure**: **2 seconds**
- **Frequency**: **60 seconds**
- **From URI**: **ping@10.33.10.49**
- **To URI**: **ping@10.33.10.56**
- Click **Finish** (Not shown)

Figure 43 – Avaya SIP Server Configuration – Heartbeat

On the **Advanced** tab, click **Edit** button to enter the following:

- Check **Enable Grooming** box
- Select **IPO** for **Interworking Profile** (see **Section 6.2.1**)
- Click **Finish** (not shown)

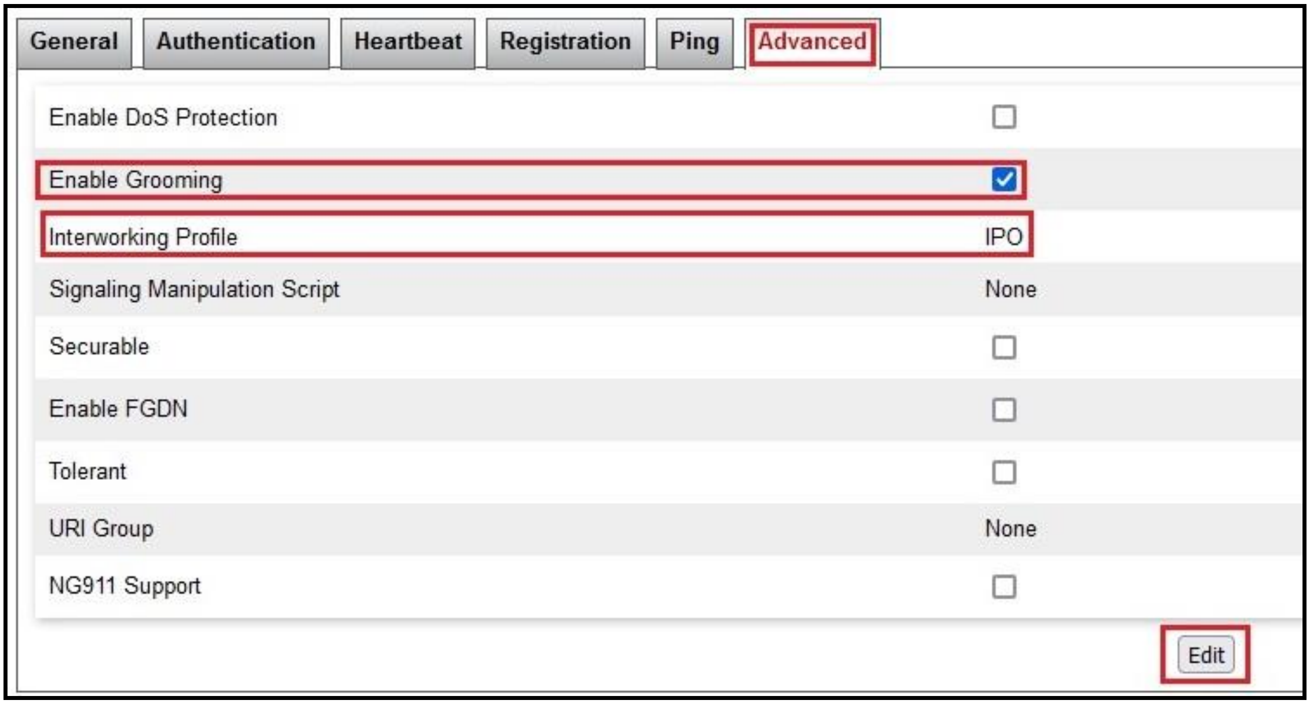


Figure 44 – Avaya SIP Server Configuration – Advanced

6.3.2. Configure SIP Server – Cox Communications

From the menu on the left-hand side, select **Services → SIP Servers → Add** and enter **Profile Name**: **SP**

On **General** tab, enter the following:

- **Server Type**: Select **Trunk Server**
- **IP Address/FQDN**: **10.33.10.60** (Cox Managed CPE LAN1 IP address)
- **Port**: **5060**
- **Transport**: **UDP**
- Click **Finish** (not shown)

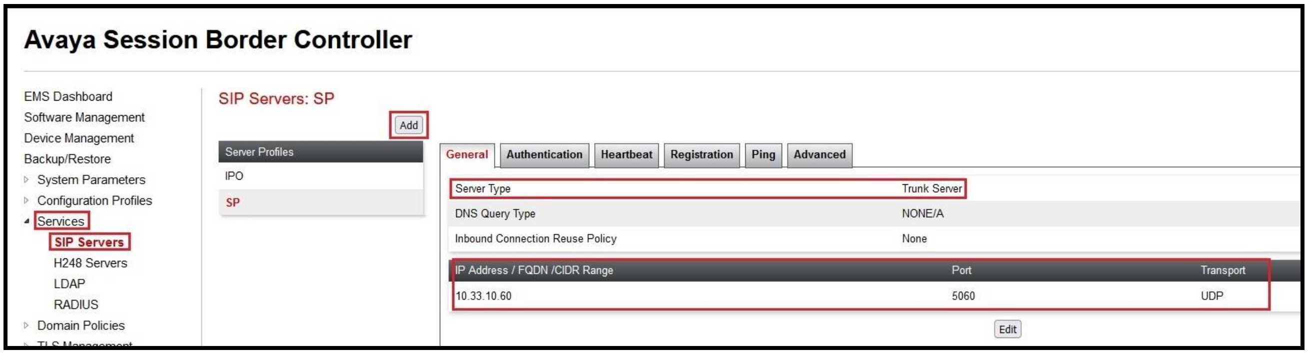


Figure 45 – Cox Communications SIP Server Configuration – General

On **Heartbeat** tab, enter the following:

- Check **Enable Heartbeat**
- Select **Method**: **OPTIONS**
- Set **Retry Timeout on Connection Failure**: **2 seconds**
- Set **Frequency**: **60 seconds**
- Input **From URI**: **402XXX6211@10.33.10.59** (Cox Communications provides this number)
- Input **To URI**: **402XXX6211@10.33.10.60** (Cox Communications provides this number)

Figure 46 - Cox Communications SIP Server – Heartbeat

On the **Advanced** tab, enter the following:

- Uncheck **Enable Grooming** option
- **Interworking Profile**: **SP** (see **Section 6.2.2**)
- Click **Finish** (not shown)

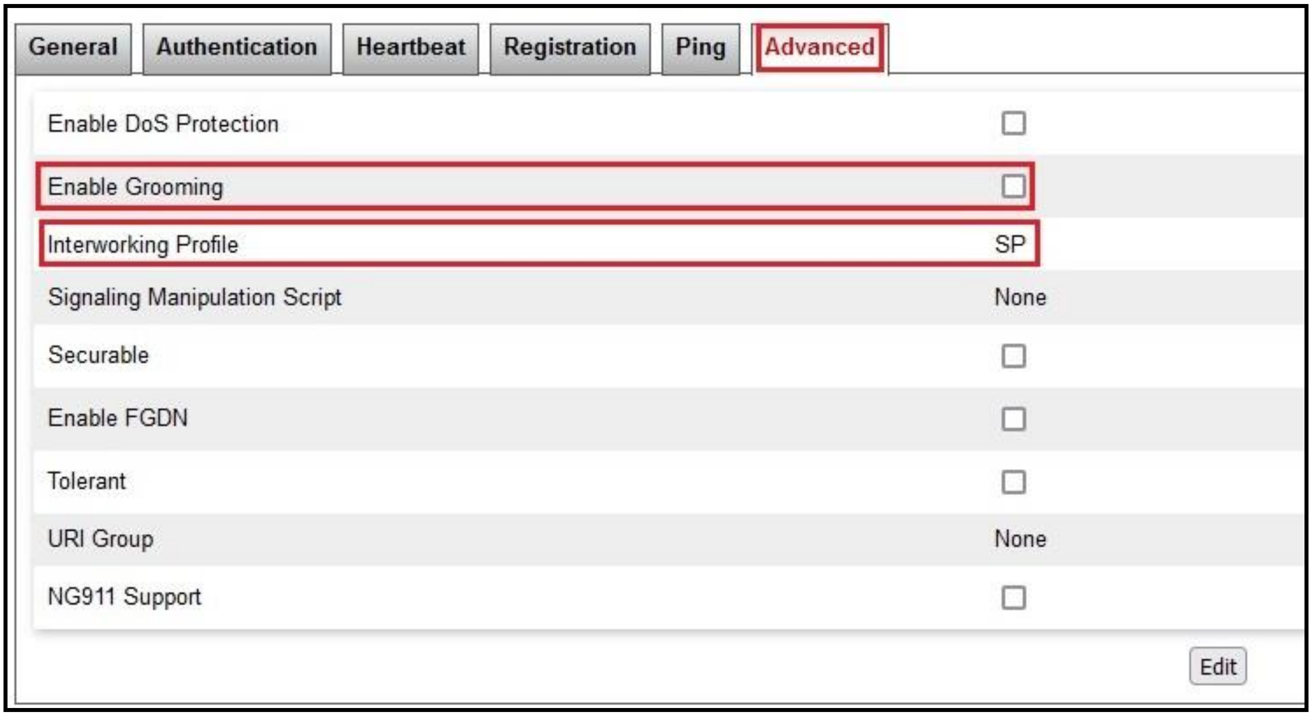


Figure 47 – Cox Communications SIP Server – Advanced

On the **Authentication** tab, enter the following:

- Check **Enable Authentication** option
- Input **User Name** (Cox Communications provides the user name)
- Leave **Realm** as blank
- Enter **Password** (Cox Communications provides the password)
- Enter **Confirm Password** (Cox Communications provides the password)
- Click **Finish**

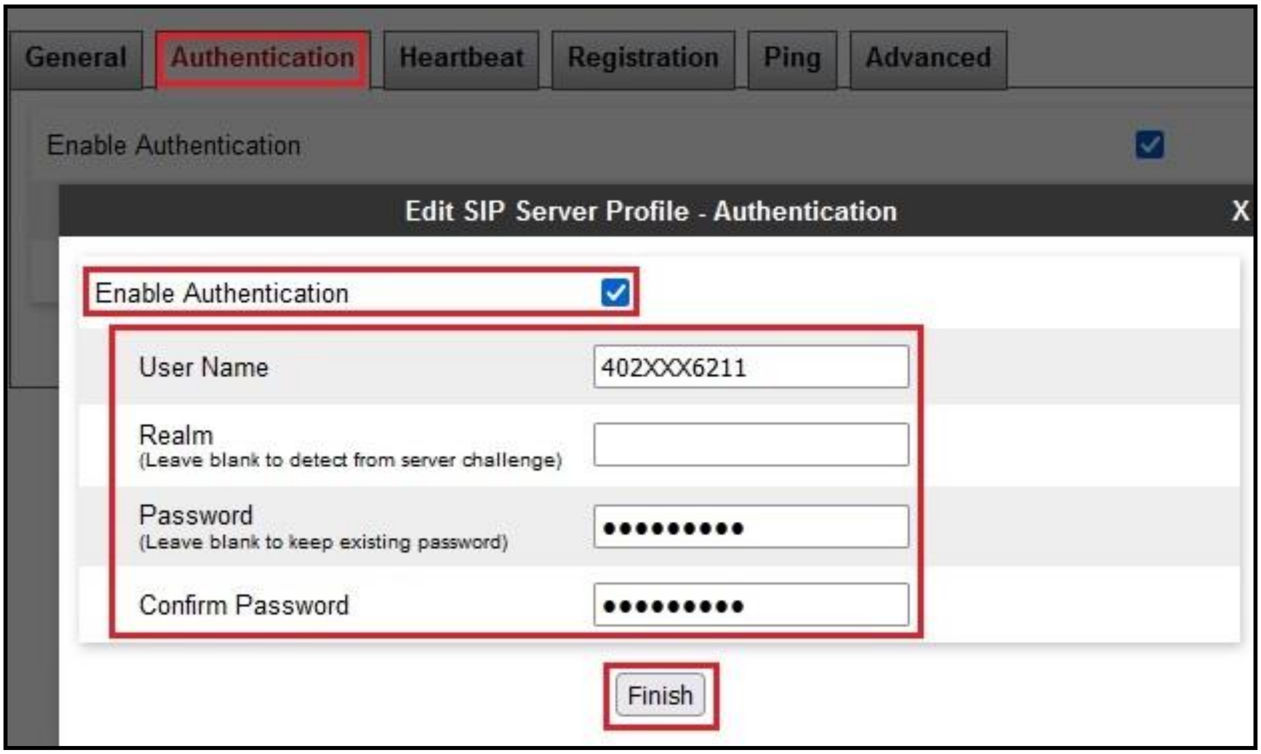


Figure 48 – Cox Communications SIP Server – Authentication

On the **Registration** tab, enter the following:

- Check **Register with All Servers** option
- Set **Refresh Interval** as **60** seconds
- Input **From URI**: **402XXX6211@10.33.10.59** (Cox Communications provides this number)
- Input **To URI**: **402XXX6211@10.33.10.60** (Cox Communications provides this number)
- Click **Finish**

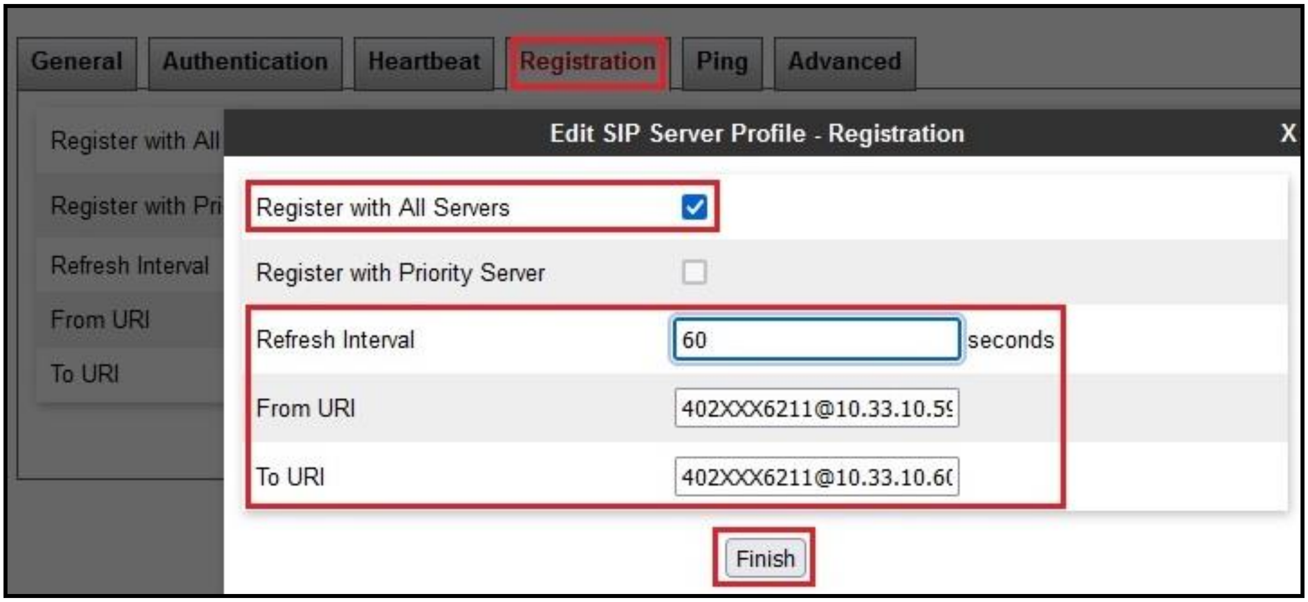


Figure 49 – Cox Communications SIP Server – Registration

6.4. Routing

Routing profiles define a specific set of routing criteria that is used, in addition to other types of domain policies, to determine the path that the SIP traffic will follow as it flows through the Avaya SBC interfaces. Two Routing Profiles were created in the test configuration, one for inbound calls, with IP Office as the destination, and the second one for outbound calls, which are routed to the service provider.

6.4.1. Configure Routing – Avaya IP Office

From the menu on the left-hand side, select **Configuration Profiles → Routing** and click **Add** as highlighted below.

Enter **Profile Name**: **To_IPO** and click **Next** button (Not shown)

- Select **Load Balancing**: **Priority**
- Check **Next Hop Priority**
- Click **Add** button to add a Next-Hop Address
- **Priority/Weight**: **1**
- **Server Configuration**: **IPO** (see **Section 6.3.1**). This selection will automatically populate the **Next Hop Address** field with **10.33.10.56:5061 (TLS)** (Avaya IP Office LAN1 port IP address)

- Click **Finish**

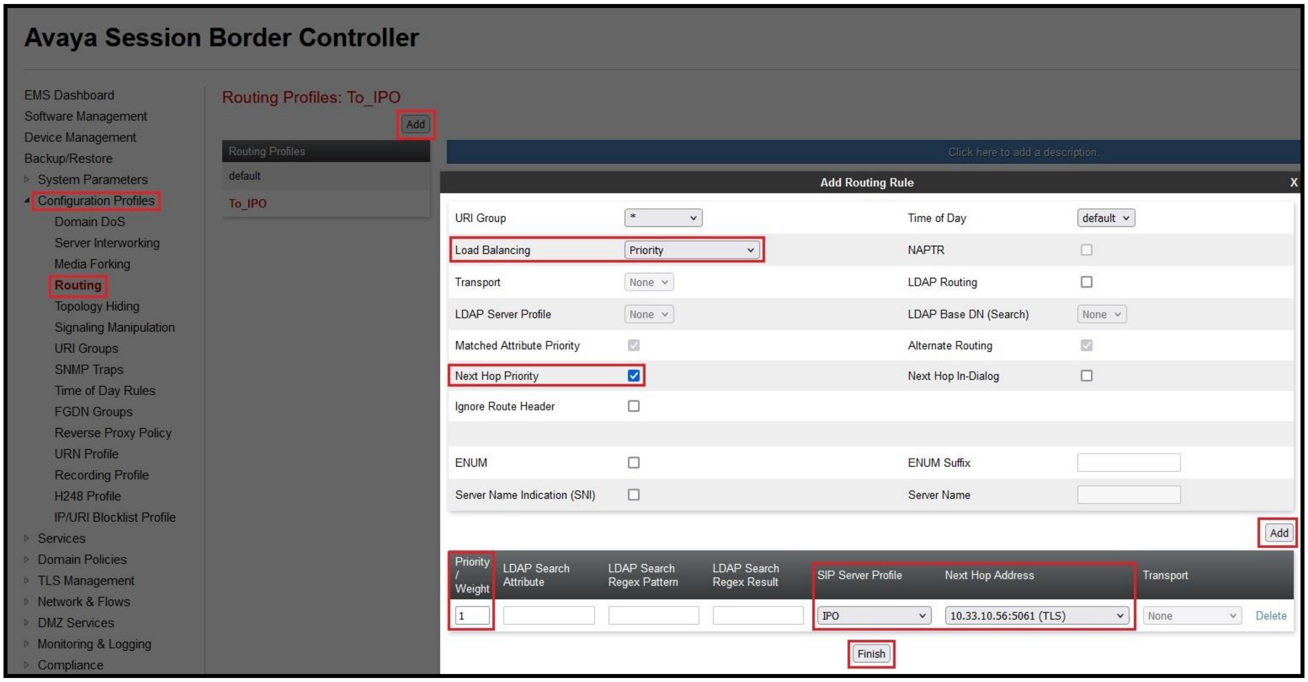


Figure 50 - Routing to Avaya IP Office

6.4.2. Configure Routing – Cox Communications

From the menu on the left-hand side, select **Configuration Profiles** → **Routing** and click **Add** as highlighted below.

Enter **Profile Name**: **To_SP** and click **Next** button (not shown)

- **Load Balancing**: **Priority**
- Check **Next Hop Priority**
- Click **Add** button to add a Next-Hop Address
- **Priority/Weight**: **1**
- **SIP Server Profile**: **SP** (see **Section 6.3.2**). This selection will automatically populate the **Next Hop Address** field with **10.33.10.60:5060 (UDP)**

- Click **Finish**

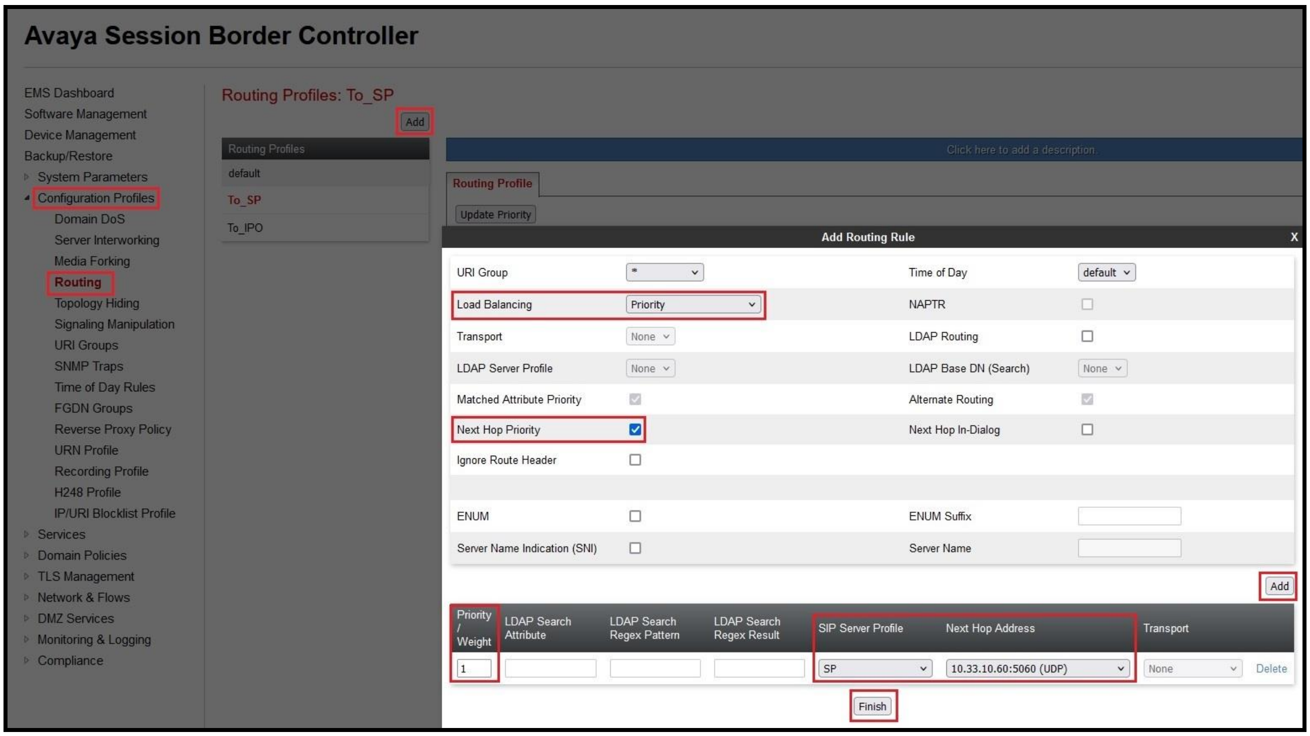


Figure 51 - Routing to Cox Communications

6.5. Configure Topology Hiding

The Topology Hiding screen allows an administrator to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks.

6.5.1. Configure Topology Hiding – Avaya Site

From the menu on the left-hand side, select **Configuration Profiles → Topology Hiding**

- Select **default** in **Topology Hiding Profiles**
- Click **Clone**
- Enter **Clone Name**: **To_IPO** and click **Finish** (not shown)
- Select **To_IPO** in **Topology Hiding Profiles** and click **Edit** button to modify as below:
  For the Header **From**,
  - In the **Criteria** column, select **IP/Domain**
  - In the **Replace Action** column, select **Overwrite**
  - In the Overwrite Value column, enter **10.33.10.49** (Avaya SBC internal IP address)

  For the Header **Request-Line**,
  - In the **Criteria** column, select **IP/Domain**
  - In the **Replace Action** column, select Overwrite
  - In the **Overwrite Value** column, enter **10.33.10.56** (Avaya IP Office LAN1 port IP address)

  For the Header **To**,
  - In the **Criteria** column, select **IP/Domain**
  - In the **Replace Action** column, select **Overwrite**
  - In the **Overwrite Value** column, enter **10.33.10.56** (Avaya IP Office LAN1 port IP address)
- Click **Finish** (not shown)

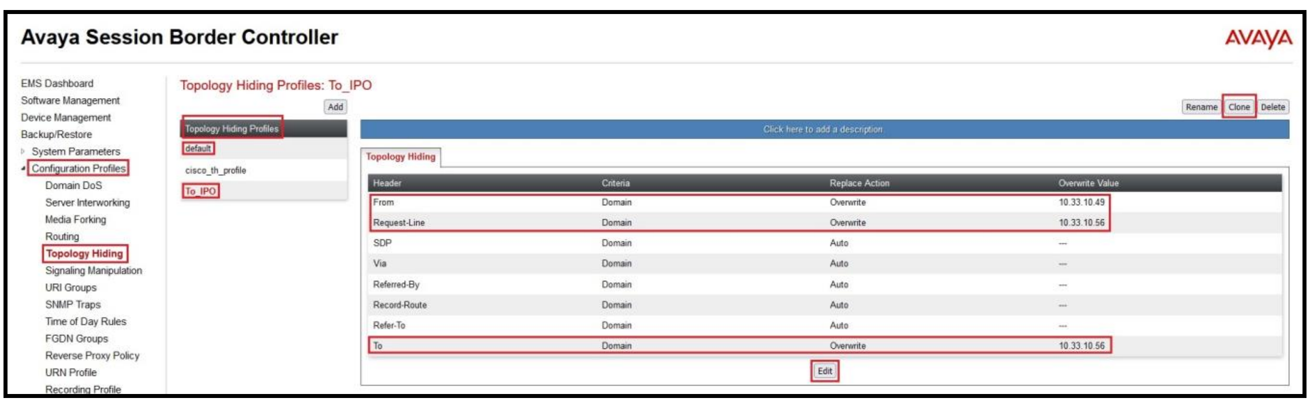


Figure 52 - Topology Hiding Avaya IP Office

6.5.2. Configure Topology Hiding – Cox Communications

From the menu on the left-hand side, select **Configuration Profiles → Topology Hiding**

- Select **default** in **Topology Hiding Profiles**
- Click **Clone**
- Enter **Clone Name**: **To_SP** and click **Finish** (not shown)
- Select **To_SP** in **Topology Hiding Profiles** and click **Edit** button to enter as below:
- For the Header **From**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **10.33.10.59** (Avaya SBC external IP address)
- For the Header **Request-Line**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **10.33.10.60** (Cox Managed CPE LAN IP address)
- For the Header **To**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **10.33.10.60** (Cox Managed CPE LAN IP address)
- Click **Finish** (not shown)

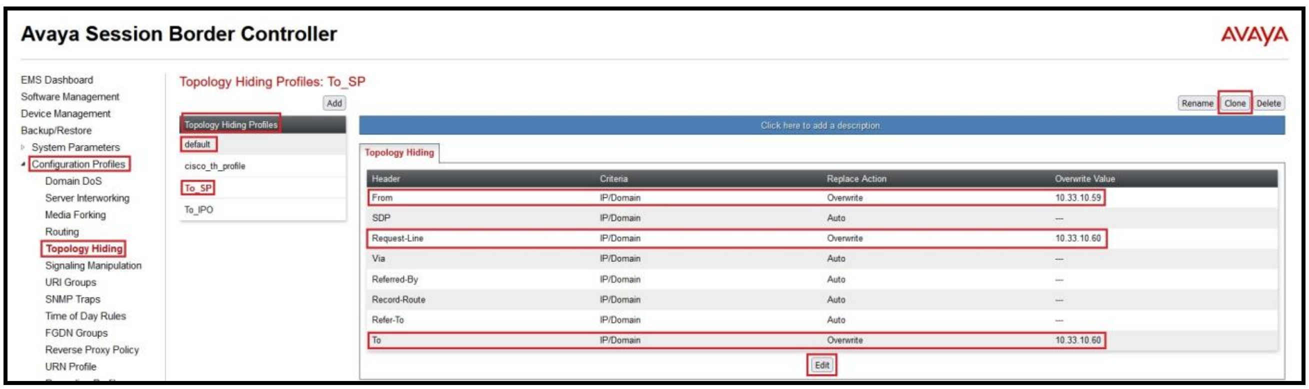


Figure 53 - Topology Hiding Cox Communications

6.6. Domain Policies

The Domain Policies feature allows administrator to configure, apply, and manage various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise. These criteria can be used to trigger different policies which will apply on call flows, change the behavior of the call, and make sure the call does not violate any of the policies. There are default policies available to use, or an administrator can create a custom domain policy.

6.6.1. Create Application Rules

Application rules define the type of SBC-based Unified Communication (UC) applications Avaya SBC protects. You can also determine the maximum number of concurrent voice and video sessions that your network can process before resource exhaustion.

From the menu on the left-hand side, select **Domain Policies → Application Rules**

- Select the **default** rule and click on **Clone** button
- Enter **Clone Name**: **App-Rules** and click **Finish** button (Not shown)
- Select the **App-Rules** rule from the list of **Application Rules** and click on **Edit** button
- Set **Maximum Concurrent Sessions** to **500** and **Maximum Sessions Per Endpoint** to **500**
- Click **Finish** button (Not shown) to save the changes

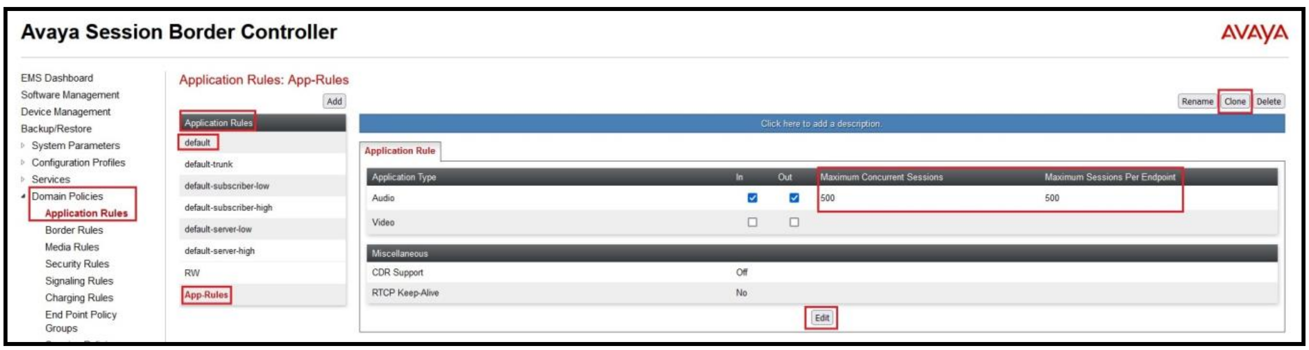


Figure 54 – Application Rule

6.6.2. Create Media Rules

Media Rules allow one to define RTP/SRTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBC security product. For the compliance test one media rule was created toward IP Office, the existing **default-low-med** media rule was used toward the Service Provider.

From the menu on the left-hand side, select **Domain Policies → Media Rules**

- Select the **avaya-low-med-enc** rule and click on **Clone** button
- Enter **Clone Name**: **IPO** and click **Finish** button (Not shown)

- Select the **IPO** rule from the list of **Media Rules** and click on **Edit** button
- Under **Audio Encryption**, select the followings:
    - **Preferred Format #1**: **SRTP_AES_CM_128_HMAC_SHA1_80**
    - **Preferred Format #2**: **RTP**
    - Check **Interworking** option
- Under **Miscellaneous**, check **Capability Negotiation**
- Click **Finish** button (Not shown) to save the changes

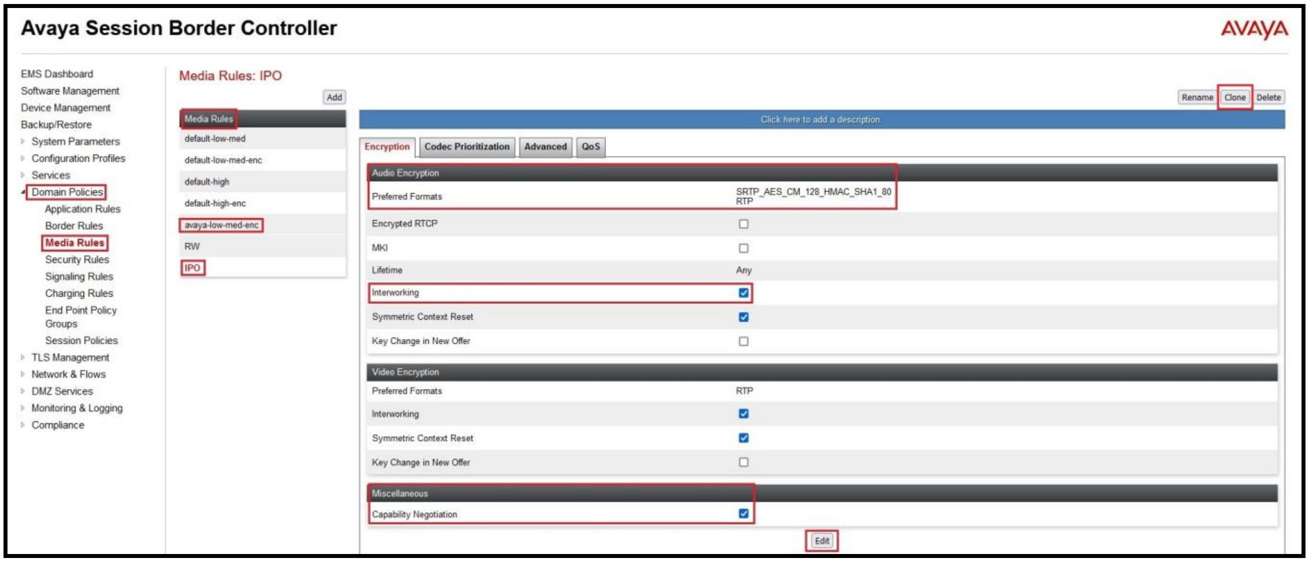


Figure 55 – Media Rule

6.6.3. Create Endpoint Policy Groups

The End-Point Policy Group feature allows one to create Policy Sets and Policy Groups. A Policy Set is an association of individual, SIP signaling-specific security policies (rule sets): application, border, media, security, and signaling, each of which was created using the procedures contained in the previous sections. A Policy Group is comprised of one or more Policy Sets. The purpose of Policy Sets and Policy Groups is to increasingly aggregate and simplify the application of Avaya SBC security features to very specific types of SIP signaling messages traversing through the enterprise.

From the menu on the left-hand side, select **Domain Policies → End Point Policy Groups**

- Select **Add**
- Enter **Group Name**: **EndPoint-Policy**
    - **Application Rule**: **App-Rules** (See **Section 6.6.1**)
    - **Border Rule**: **default**
    - **Media Rule**: **IPO** (See **Section 6.6.2**)
    - **Security Rule**: **default-low**
    - **Signaling Rule**: **default**
    - Leave other options as default
- Select **Finish** (not shown)

Figure 56 – End Point Policy – IPO

From the menu on the left-hand side, select **Domain Policies → End Point Policy Groups**

- Select **Add**
- Enter **Group Name**: **SP**
  - **Application Rule**: **App-Rules** (See **Section 6.6.1**)
  - **Border Rule**: **default**
  - **Media Rule**: **default-low-med**
  - **Security Rule**: **default-low**
  - **Signaling Rule**: **default**
  - Leave other options as default
- Select **Finish** (not shown)

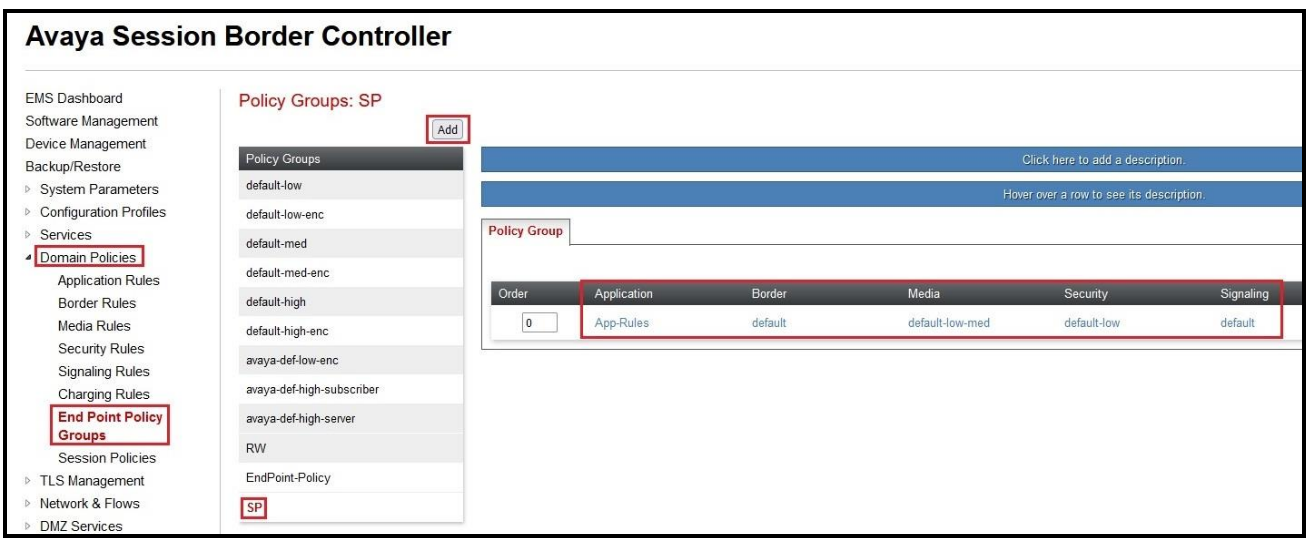


Figure 57 – End Point Policy – Cox Communications

6.7. Network & Flows

The Network & Flows feature for SIP allows one to view aggregate system information and manage various device-specific parameters which determine how a particular device will function when deployed in the network.

6.7.1. Manage Network Settings

From the menu on the left-hand side, select **Network & Flows → Network Management**.

- Select **Networks** tab and click the **Add** button to add a network for the inside interface as follows:
  - **Name**: **Network_A1**
  - **Default Gateway**: **10.33.10.1**
  - **Subnet Mask**: **255.255.255.0**
  - **Interface**: **A1** (This is the Avaya SBC inside interface)
  - Click the **Add** button to add the **IP Address** for inside interface: **10.33.10.49**
  - Click the **Finish** button to save the changes

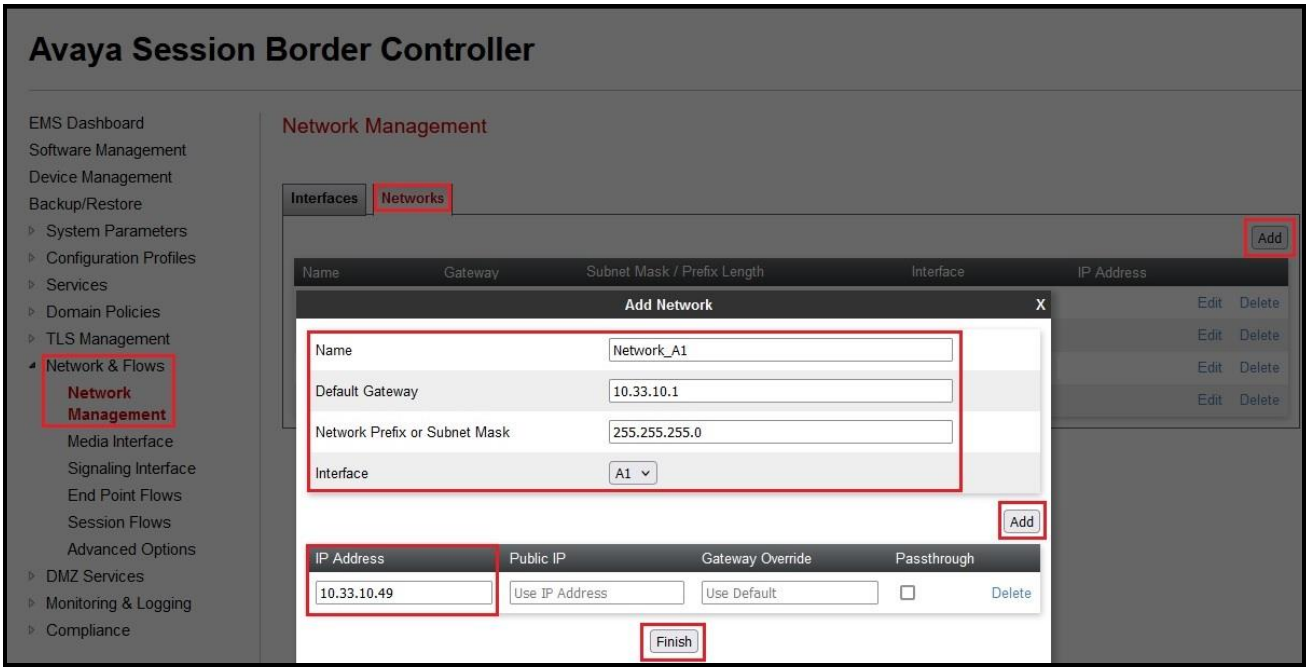


Figure 58 - Network Management – Inside Interface

From the menu on the left-hand side, select **Network & Flows → Network Management**.

- Select **Networks** tab and click **Add** button to add a network for the outside interface as follows:
  - **Name**: **Network_B1**
  - **Default Gateway**: **10.33.10.1**
  - **Subnet Mask**: **255.255.255.0**
  - **Interface**: **B1** (This is the Avaya SBC outside interface)
  - Click the **Add** button to add the **IP Address** for outside interface: **10.33.10.59**
  - Click the **Finish** button to save the changes

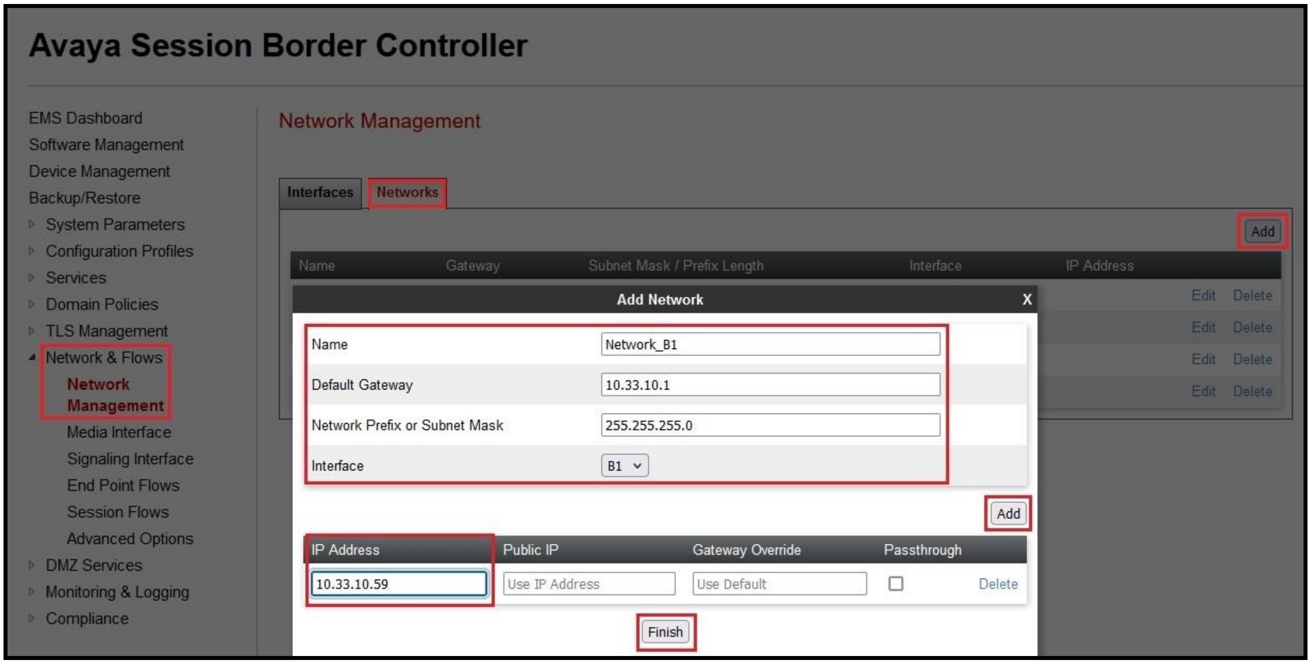


Figure 59 - Network Management – Outside Interface

From the menu on the left-hand side, select **Network & Flows → Network Management**

- Select the **Interfaces** tab
- Click on the **Status** of the physical interfaces being used and change them to **Enabled** state

-

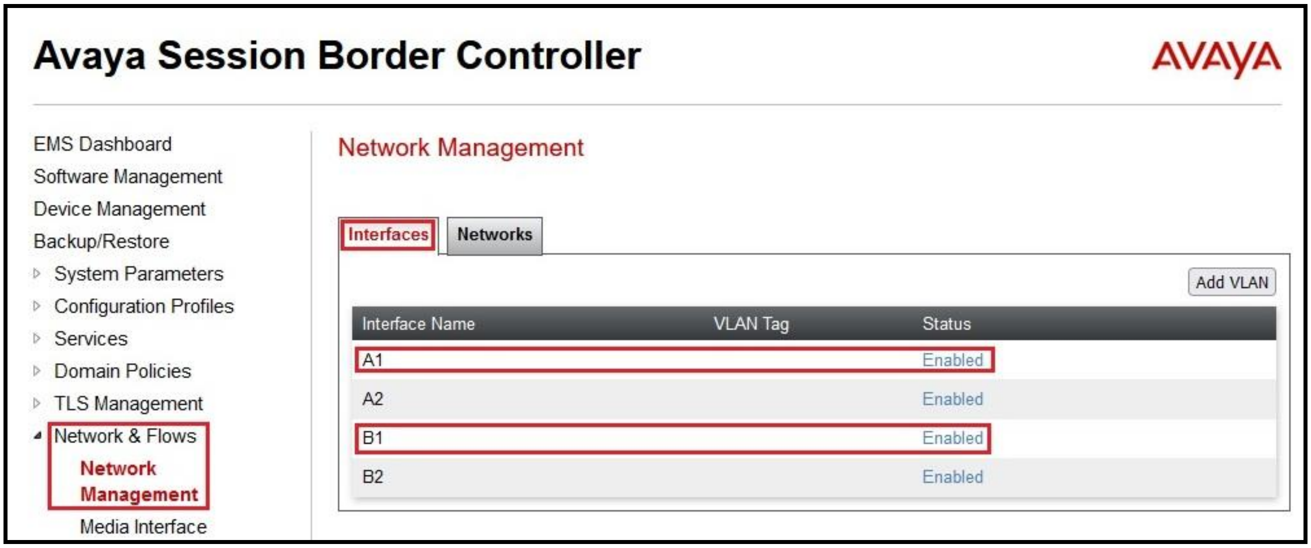


Figure 60 - Network Management – Interface Status

6.7.2. Create Media Interfaces

Media Interfaces define the IP Addresses and port ranges in which the Avaya SBC will accept media streams on each interface. The default media port range on the Avaya SBC can be used for both inside and outside ports.

From the menu on the left-hand side, **Network & Flows → Media Interface**

- Select the **Add** button and enter the following:
  - **Name**: **InsideMedia**
  - **IP Address**: Select **Network_A1 (A1, VLAN 0)** and **10.33.10.49** (Internal IP address toward IP Office)
  - **Port Range**: **35000 – 40000**
  - Click **Finish** (not shown)
- Select the **Add** button and enter the following:
  - **Name**: **OutsideMedia**
  - **IP Address**: Select **Network_B1 (B1, VLAN 0)** and **10.33.10.59** (External IP address toward Cox Managed CPE)
  - **Port Range**: **35000 – 40000**
  - Click **Finish** (not shown)

Figure 61 - Media Interface

6.7.3. Create Signaling Interfaces

Signaling Interfaces define the type of signaling on the ports.

From the menu on the left-hand side, select **Network & Flows → Signaling Interface**

- Select the **Add** button and enter the following:
  - **Name**: **OutsideSig**
  - **IP Address**: Select **Network_B1 (B1, VLAN 0)** and **10.33.10.59** (External IP address toward Cox Managed CPE)
  - **UDP Port**: **5060**
  - Click **Finish** (not shown)

From the menu on the left-hand side, select **Network & Flows → Signaling Interface**

- Select the **Add** button and enter the following:
  - **Name**: **InsideSig**
  - **IP Address**: Select **Network_A1 (A1, VLAN 0)** and **10.33.10.49** (Internal IP address toward IP Office)
  - **TLS Port**: **5061**
  - **TLS Profile**: **AvayaSBCServer**. Note: During the compliance test in the lab environment, demo certificates are used on Session Manager, and are not recommended for production use.
  - Click **Finish** (not shown)

Note: For the external interface, the Avaya SBC was configured to listen for UDP on port 5060 the same as Cox Communications used. For the internal interface, the Avaya SBC was configured to listen for TLS on port 5061.

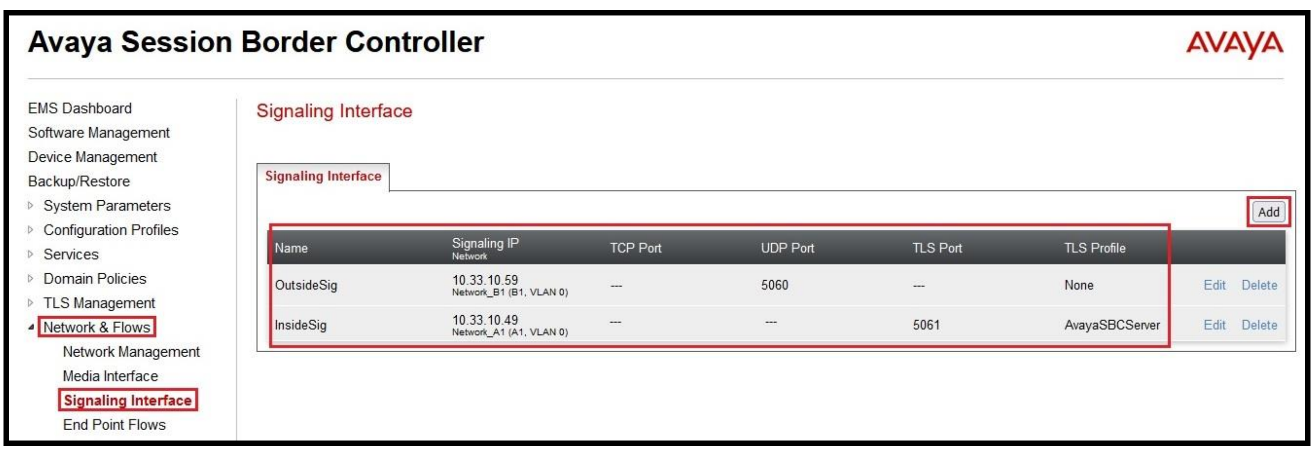


Figure 62 - Signaling Interface

6.7.4. Configure Server Flows

Server Flows allow an administrator to categorize signaling and apply various policies.

6.7.4.1 Create End Point Flows – Avaya IP Office

From the menu on the left-hand side, select **Network & Flows → End Point Flows**

- Select the **Server Flows** tab
- Select **Add**, enter the followings:
  - **Flow Name**: **IPO Flow**
  - **Server Configuration**: **IPO** (see **Section 6.3.1**)
  - **URI Group**: *****
  - **Transport**: *****
  - **Remote Subnet**: *****
  - **Received Interface**: **OutsideSig** (see **Section 6.7.3**)
  - **Signaling Interface**: **InsideSig** (see **Section 6.7.3**)
  - **Media Interface**: **InsideMedia** (see **Section 6.7.2**)
  - **Secondary Media Interface**: **None**
  - **End Point Policy Group**: **EndPoint-Policy** (see **Section 6.6.3**)
  - **Routing Profile**: **To_SP** (see **Section 6.4.2**)
  - **Topology Hiding Profile**: **To_IPO** (see **Section 6.5.1**)
  - Leave other options as default
  - Click **Finish** to save the changes

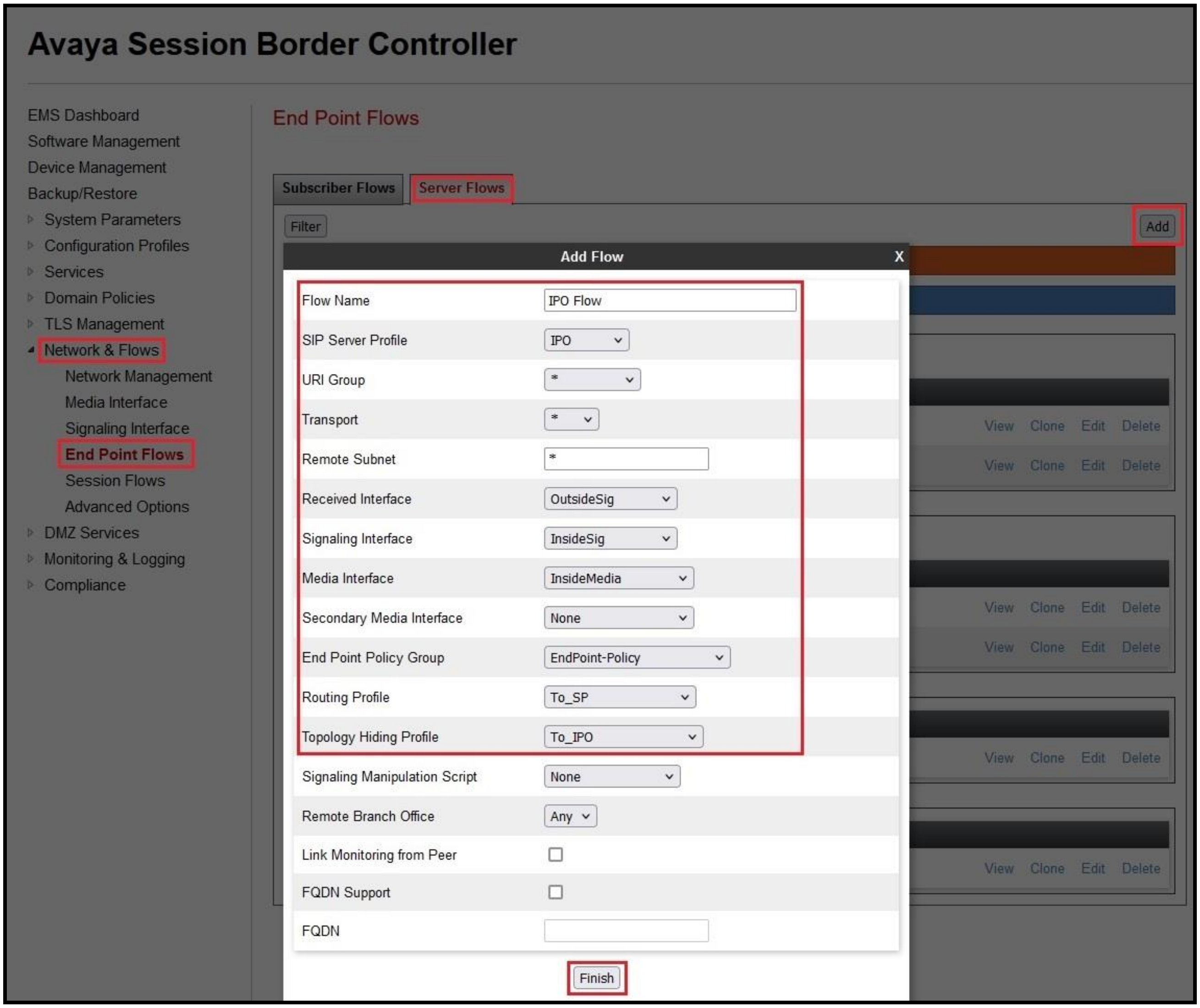


Figure 63 - End Point Flow 1

6.7.4.2 Create End Point Flows – Cox Communications

From the menu on the left-hand side, select **Network & Flows → End Point Flows**

- Select the **Server Flows** tab
- Select **Add**, enter the followings:
  - **Flow Name**: **SP Flow**
  - **Server Configuration**: **SP** (see **Section 6.3.2**)
  - **URI Group**: *****
  - **Transport**: *****
  - **Remote Subnet**: *****
  - **Received Interface**: **InsideSig** (see **Section 6.7.3**)
  - **Signaling Interface**: **OutsideSig** (see **Section 6.7.3**)
  - **Media Interface**: **OutsideMedia** (see **Section 6.7.2**)
  - **Secondary Media Interface**: **None**
  - **End Point Policy Group**: **SP** (see **Section 6.6.3**)
  - **Routing Profile**: **To_IPO** (see **Section 6.4.1**)
  - **Topology Hiding Profile**: **To_SP** (see **Section 6.5.2**)
  - Leave other options as default
  - Click **Finish** to save the changes

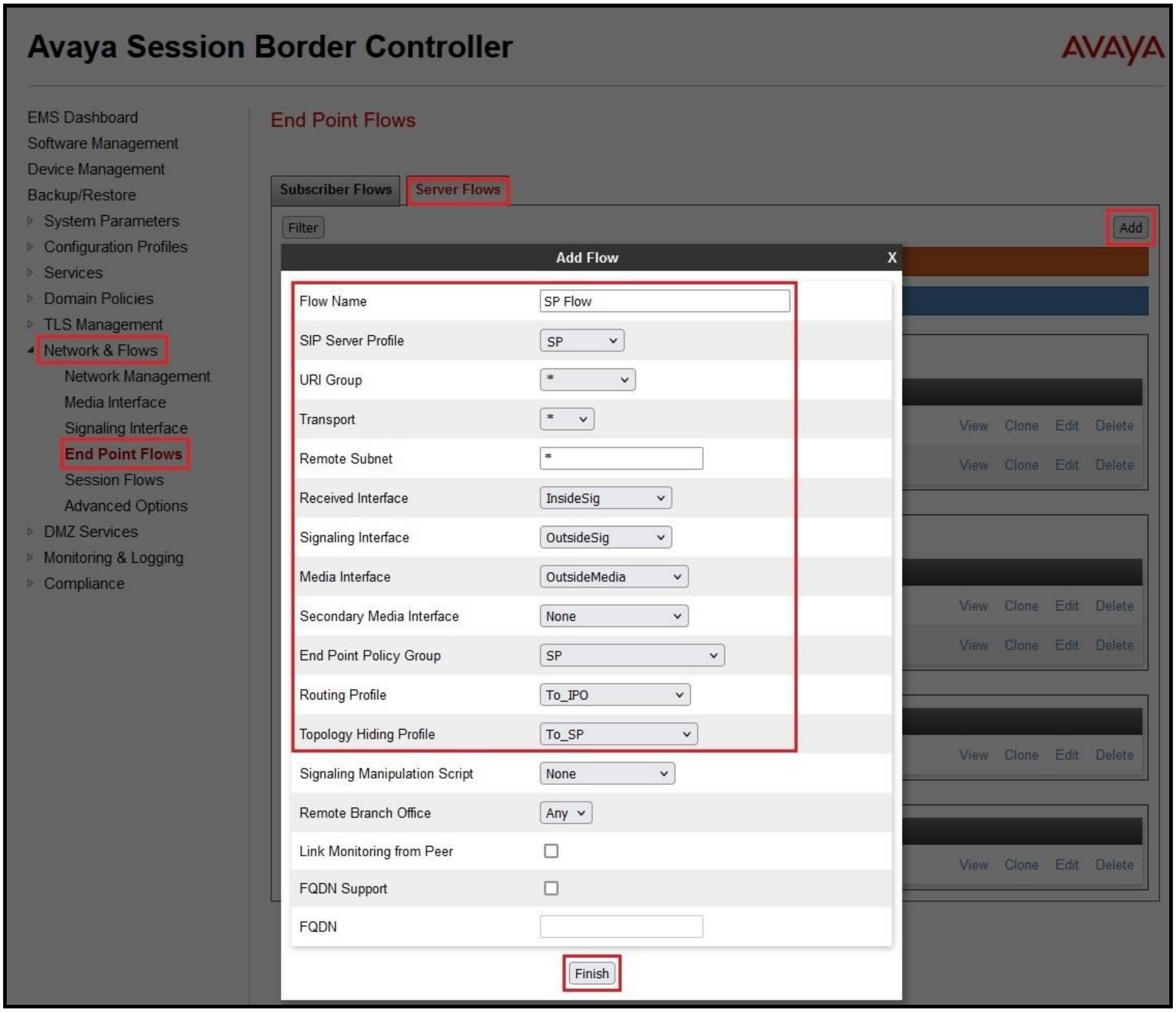


Figure 64 - End Point Flow 2

7. Cox Communications SIP Trunk Configuration

Cox Communications is responsible for the configuration of Cox Communications SIP Trunk Service. Cox Communications will provide the Cox Managed CPE to the customer when the customer orders the Cox Communications SIP trunk service. Cox Communications will be responsible for managing the Cox Managed CPE. Customer must provide the IP Address used to reach the Avaya SBC public interface at the enterprise. Cox Communications will provide the customer necessary information to configure the SIP connection between Avaya SBC and Cox Managed CPE. Cox Communications also provides the Cox Communications SIP Specification document for reference.

The configuration between Cox Communications SIP Trunk and the enterprise is a static IP Address configuration.

8. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office Monitor application to monitor the active SIP call traces between the enterprise and Cox Communications. Launch the application from **Start → All apps → IP Office → Monitor** on the PC where Avaya IP Office Server Edition Manager was installed. Click start/ stop buttons to capture the SIP call traces.

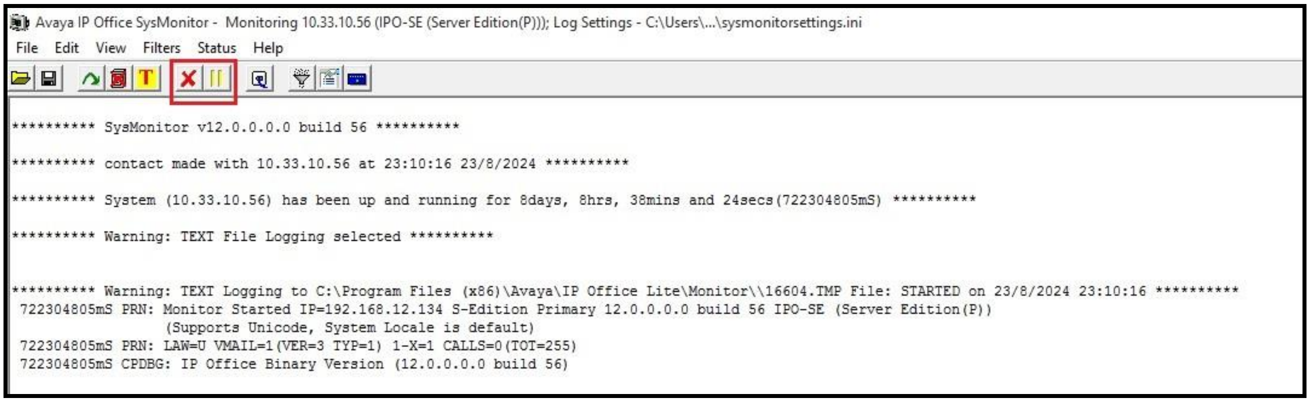


Figure 65 – SIP Trace Monitor

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP Line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** for each channel. (The below screen shot showed two active calls at the time)

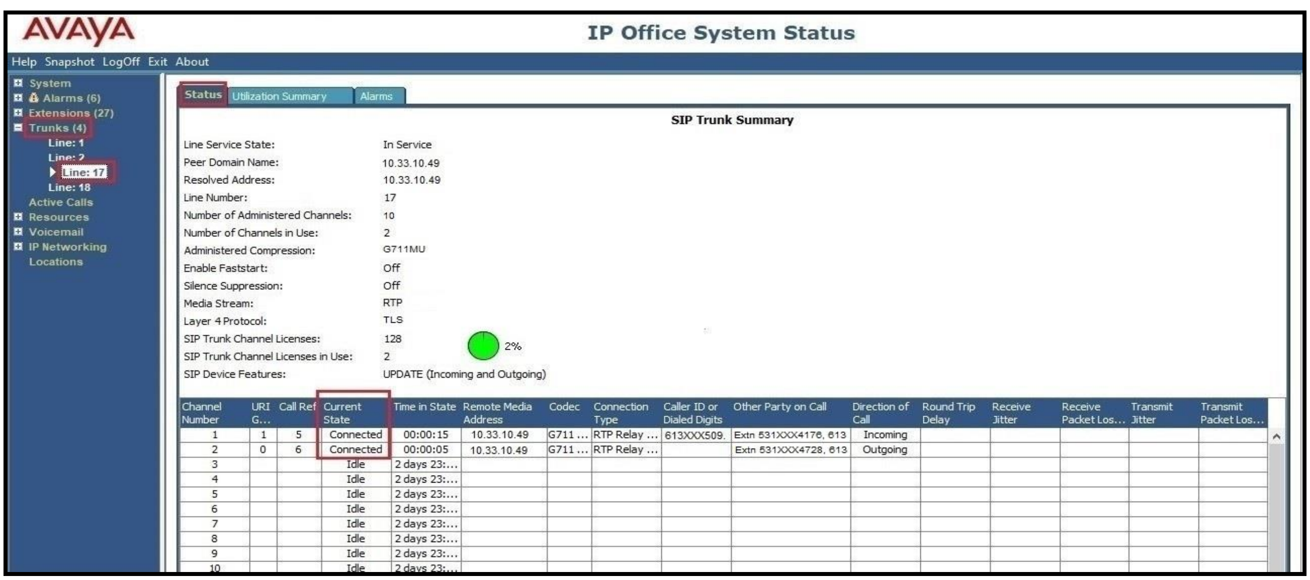


Figure 66 – SIP Trunk status

- Use the Avaya IP Office System Status application to verify that no alarms are active on the SIP line. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select **Alarm → Trunks** to verify that no alarms are active on the SIP line.

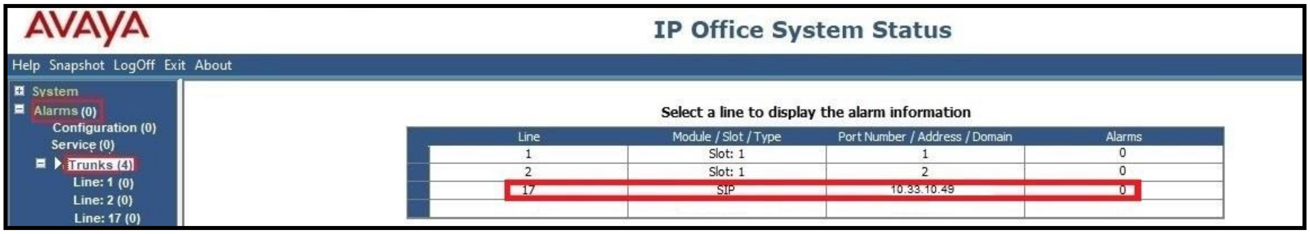


Figure 67 – SIP Trunk alarm

- Verify that a phone connected to the PSTN can successfully place a call to Avaya IP Office with two-way audio.
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Capture SIP call traces on Avaya SBC by executing command via the Command Line Interface (CLI): Login Avaya SBC with root user and enter the command: #traceSBC. The tool updates the database directly based on which trace mode is selected.

9. Conclusion

Cox Communications successfully passed compliance testing via the Avaya DevConnect Program. These Application Notes describe the procedures required to configure SIP trunk connectivity between Avaya IP Office 12.0 and the Avaya SBC 10.2 to support Cox Communications SIP Trunking service, as shown in **Figure 1**.

10. Additional References

[1] Avaya IP Office™ Platform Release 12.0 Release Notes / Technical Bulletin General Availability Issue 002, June 20th, 2024

[2] Deploying IP Office Server Edition and Application Servers, Release 12.0 Issue 31, April 2024

[3] Deploying Avaya IP Office Servers as Virtual Machines, Release 12.1, Issue 23, August 2024

[4] IP Office™ Platform 12.0, Deploying an IP Office 500 V2/V2A in IP Office Basic Edition Mode, Issue 41e, May 29th, 2024

[5] Administering Avaya IP Office using Manager, Release 12.0, Issue 51.1.2, June 2024.

[6] Deploying Avaya Session Border Controller on a Virtualized Environment Platform, Release 10.2.x, Issue 1, June 2024.

[7] Administering Avaya Session Border Controller, Release 10.2.x, Issue 3, July 2024.

[8] Application Notes for Configuring Remote Workers with Avaya Session Border Controller 8.1 on the Avaya Aura® Platform – Issue 1.0

Product documentation for Avaya products may be found at: http://support.avaya.com.

Additional IP Office documentation can be found at:
https://ipofficekb.avaya.com/businesspartner/index.html

Product documentation for Cox Communications SIP Trunking may be found at:
http://www.cox.com

11. Appendix - Cox Managed CPE Configuration

The Cox Managed CPE is configured to manage all SIP signaling and provides voice quality management. All data traffic also traverses the Cox Managed CPE. It is part of the Cox Communications SIP trunk service and Cox Communications will provide it to the customer when the customer orders the Cox Communications SIP trunk service. Cox Communications manages it and the end-customer does not manage.

Note: Cox Managed CPE is part of Cox Communications SIP trunk service offering and it is Cox Communications' responsibility for all the aspect of the Cox Managed CPE (i.e., support, detail configuration, maintenance, etc.). The Cox Managed CPE's sample configuration included in this document is used during this compliance testing.

11.1. Cox Managed CPE Login

The Cox Managed CPE was configured with a local LAN address of 10.33.10.60 and a subnet mask of 255.255.255.0. A personal computer is configured with Ethernet IP address assigned to any address other than 10.33.10.49 in the same subnet mask, for example 10.33.10.60
Launch a web browser on personal computer and enter the following URL: http://10.33.10.60 and hit enter.

The following login window should appear:

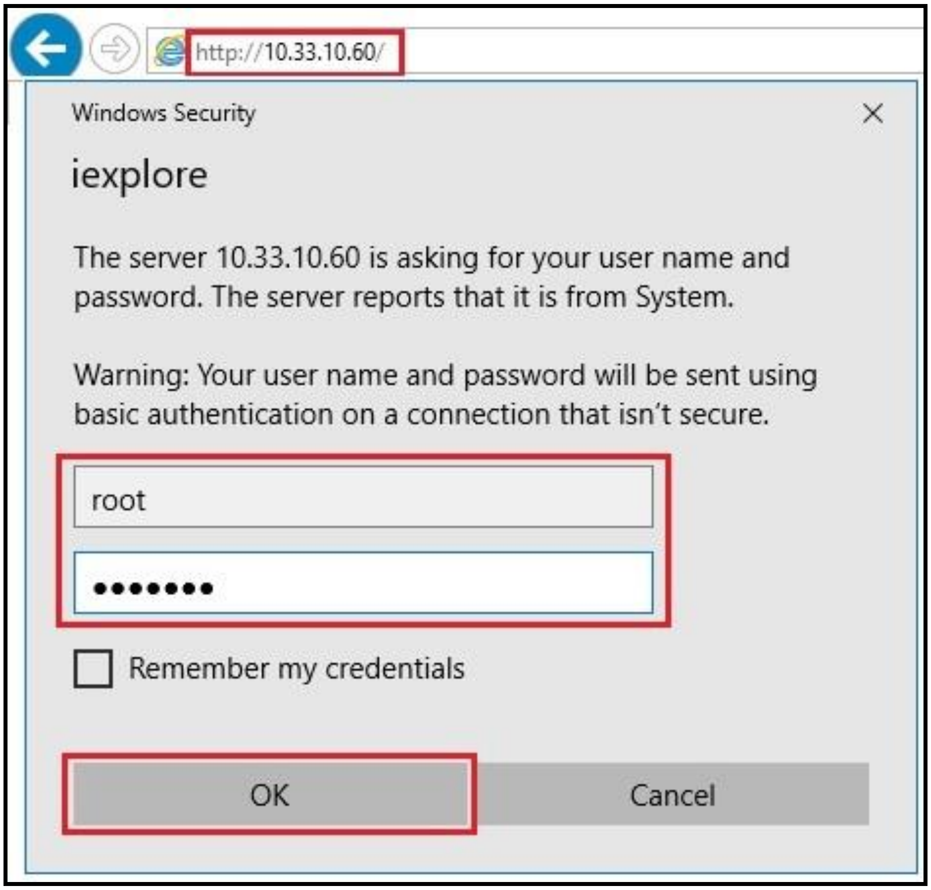


Figure 68 – Cox Managed CPE Login

- Enter **User Name** and **Password** field
- Click **OK** and the system page should be appeared next

11.2. Network Configuration

From the Configuration Menu, select Network menu option.
Under Network, input the public and private networks as followings:

- LAN Interface Settings:
  - **IP Address**: **10.33.10.60**
  - **Subnet Mask**: **255.255.255.0**
  - Check **Enable VLAN Support**
  - **Default VLAN ID**: **1**
- WAN Interface IPv4 Settings:
  - Check **Static IP**
  - **IP Address**: **10.10.80.103** (Provide this IP Address to service provider to set up the connectivity)
  - **Subnet Mask**: **255.255.255.128**
- Network Settings:
  - **Default Gateway**: **10.10.80.1**

Submit the changes.

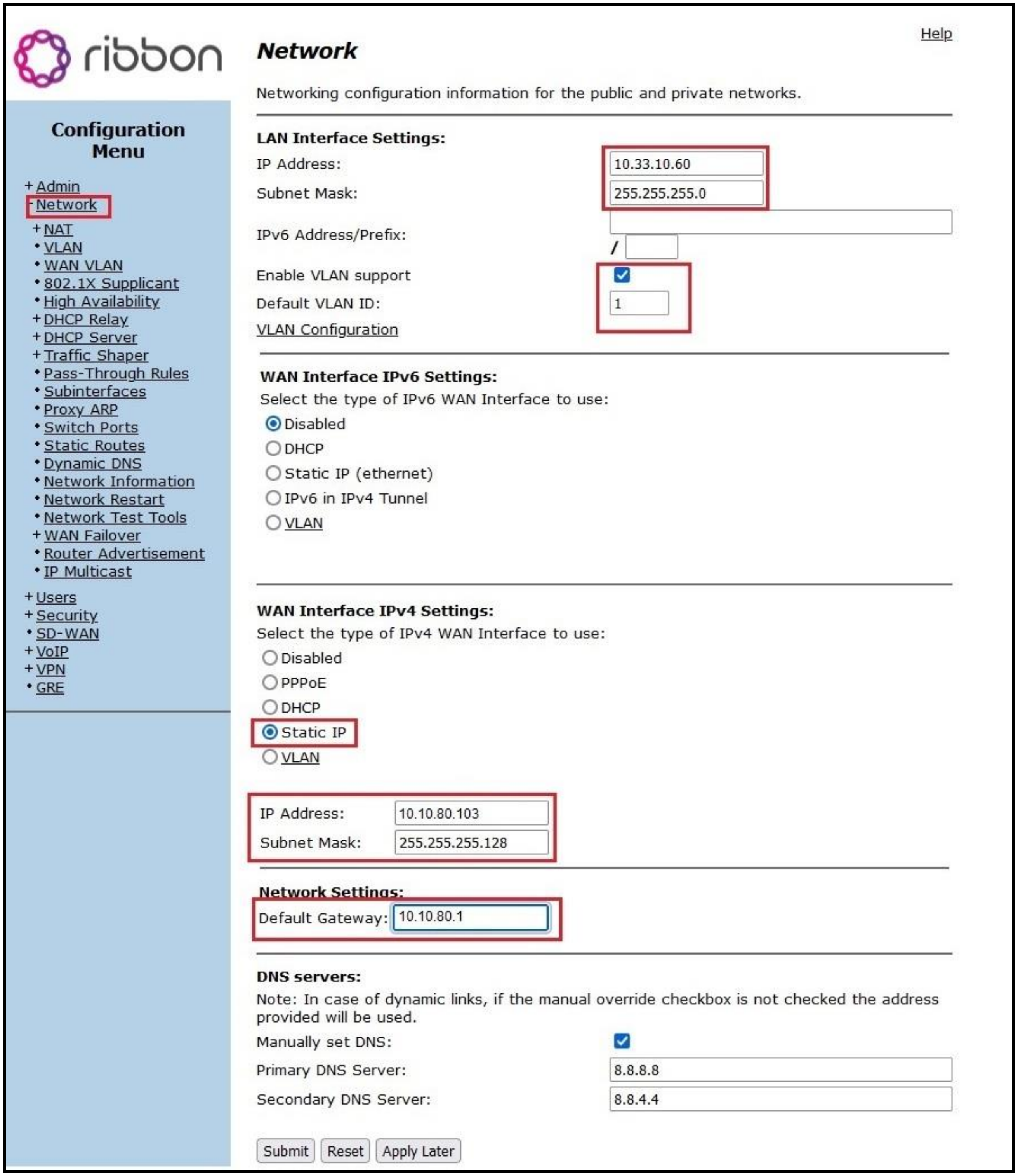


Figure 69 – Cox Managed CPE Network Configuration

11.3. VLAN Configuration

There is a VLAN which has been created and configured as shown in capture below. Details how to create the VLAN is not shown.

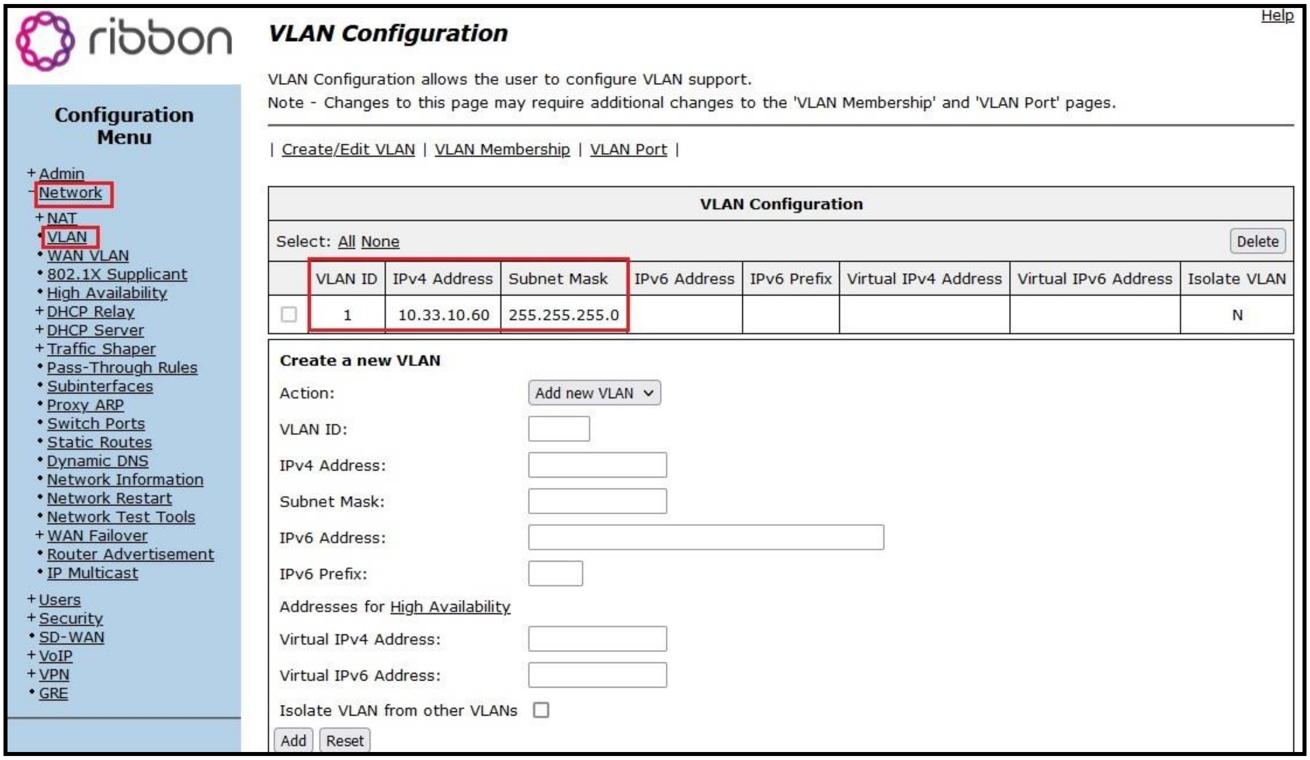


Figure 70 – Cox Managed CPE VLAN Configuration

11.4. VoIP Settings

From the **Configuration Menu**, select **VoIP** menu option → **SIP** option.
Under **SIP Settings**, input the parameters as followings:

- **SIP Server Address**: **DUKEBWSSCM-MTC1-SA-XXXXXX.TC.AT.COX.NET**
- **SIP Server Port**: **5060**
- **SIP Server Transport: Pass Through**
- Check **Use Custom Domain**
- **SIP Server Domain**: **coxbusiness.com**

Submit the changes.

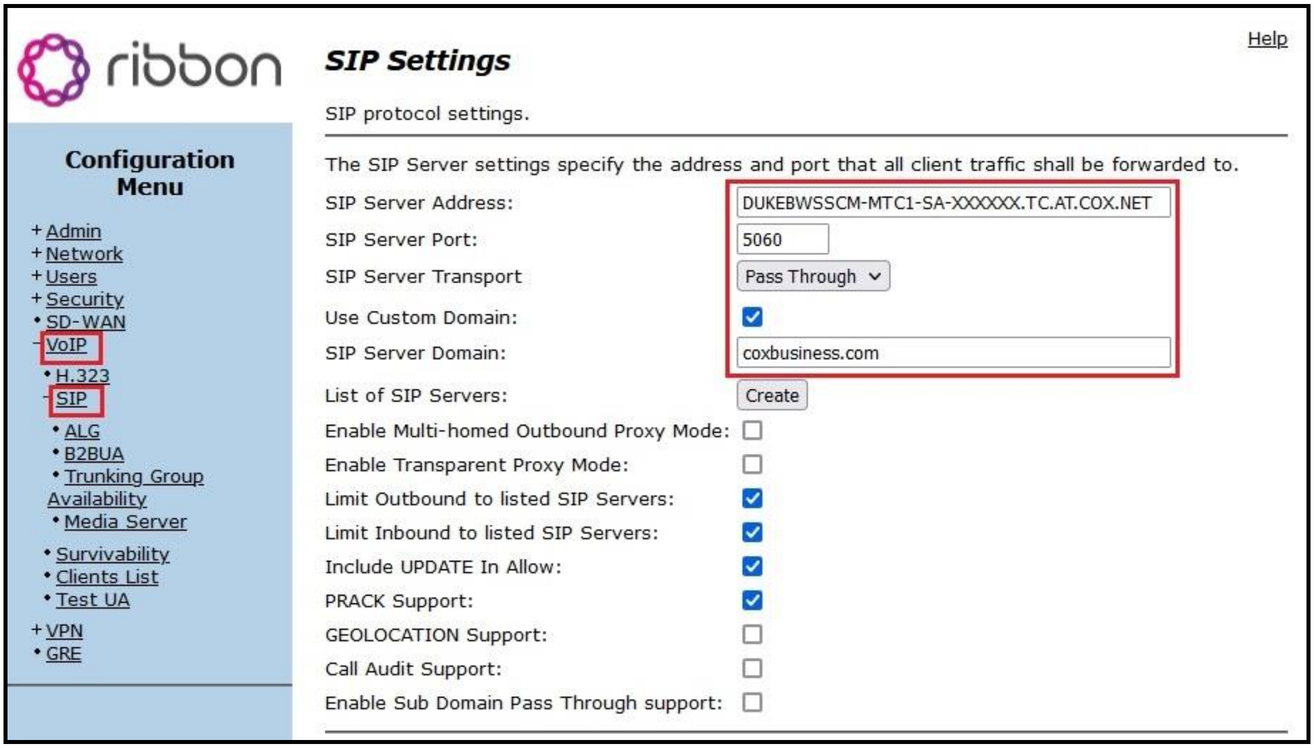


Figure 71 – Cox Managed CPE VoIP Settings

From the **Configuration Menu**, select **Survivability** to check SIP Server Reachability status. When the SIP Server connectivity is up, the status is Active.

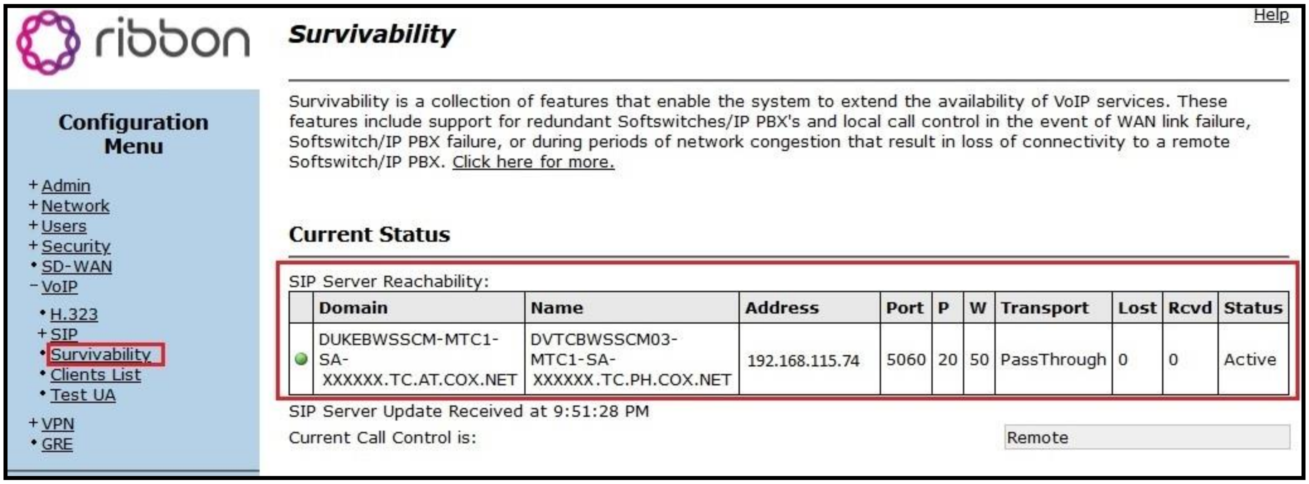


Figure 72 – Cox Managed CPE SIP Server Survivability

11.5. B2BUA Trunking Configuration

From the **Configuration Menu**, select **VoIP** menu option → **SIP** → **B2BUA**.
Under **Trunking Devices**:

- Input a recognizable **Name** for the trunking device: **Avaya**
- At **Model** pull down menu, choose **Avaya IP Office**
- Input **IP Address** of the Avaya IP Office server: **10.33.10.59**
- Input SIP **Port** of the Avaya IP Office: **5060**
- Select **Transport** as **UDP**
- Input **Username**: **402XXX6211**, which is pilot number for trunk registration to Cox Communications system
- Input **Password**: **xxxxxxxxxxx**, which is provided by Cox Communications

Select **Update** button to create trunking device.

Under **Credentials and Registration**:

- Input **Username** as **402XXX6211**
- Input **Auth-User** as **402XXX6211**
- Input **Password** and **Confirm Password**: **xxxxxxxxxxx**, same as Trunking Devices session above

Select **Update** button.

When the trunk is successfully registered to Cox Communications system, **Status** will be shown as **OK**.

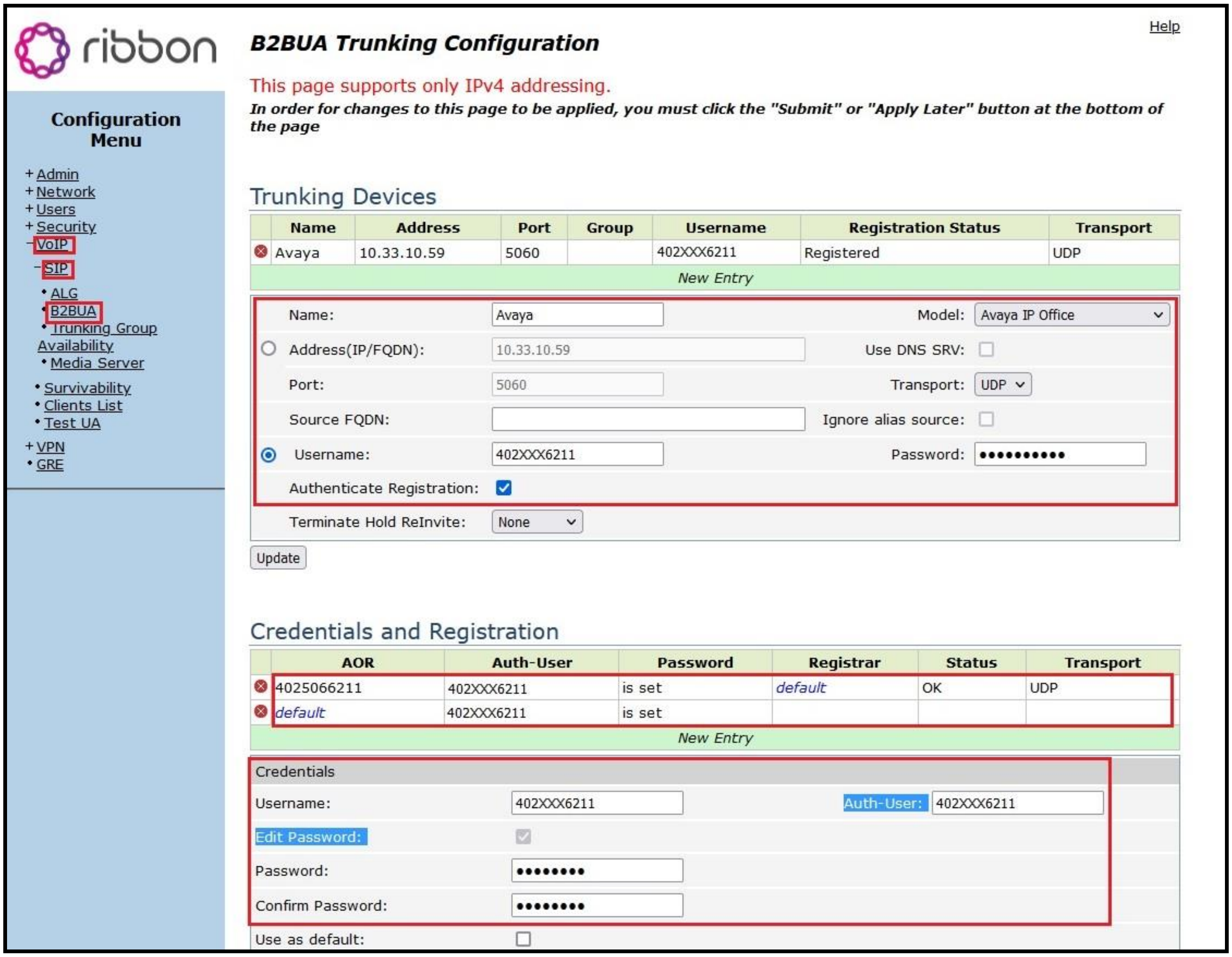


Figure 73 – Cox Managed CPE SIP Trunk Configuration

The following captured screens show the rest of the B2BUA Trunking Configuration page, continue from above screen. Detail configuration is not discussed here.

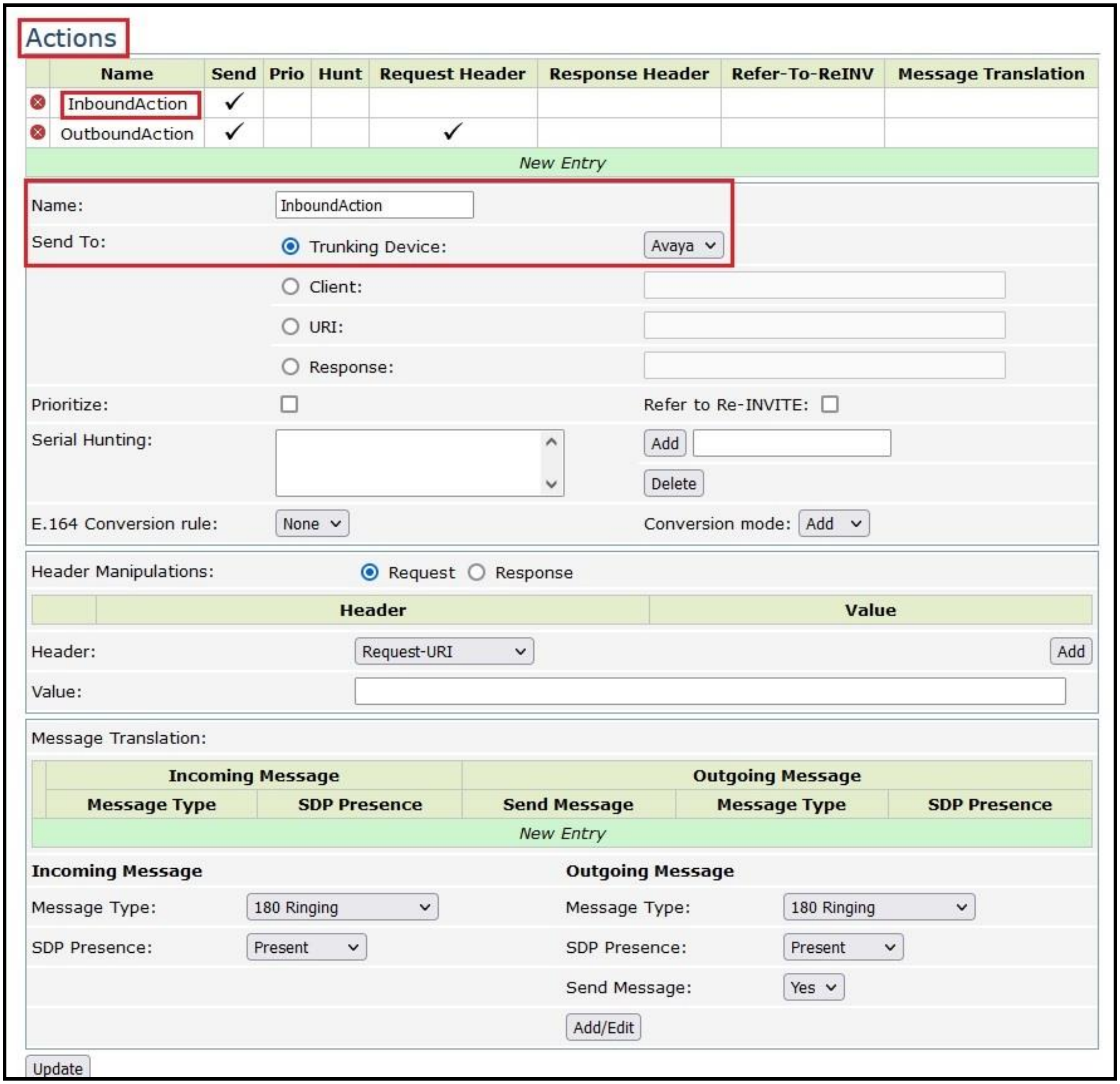


Figure 74 – Cox Managed CPE Inbound Action Configuration

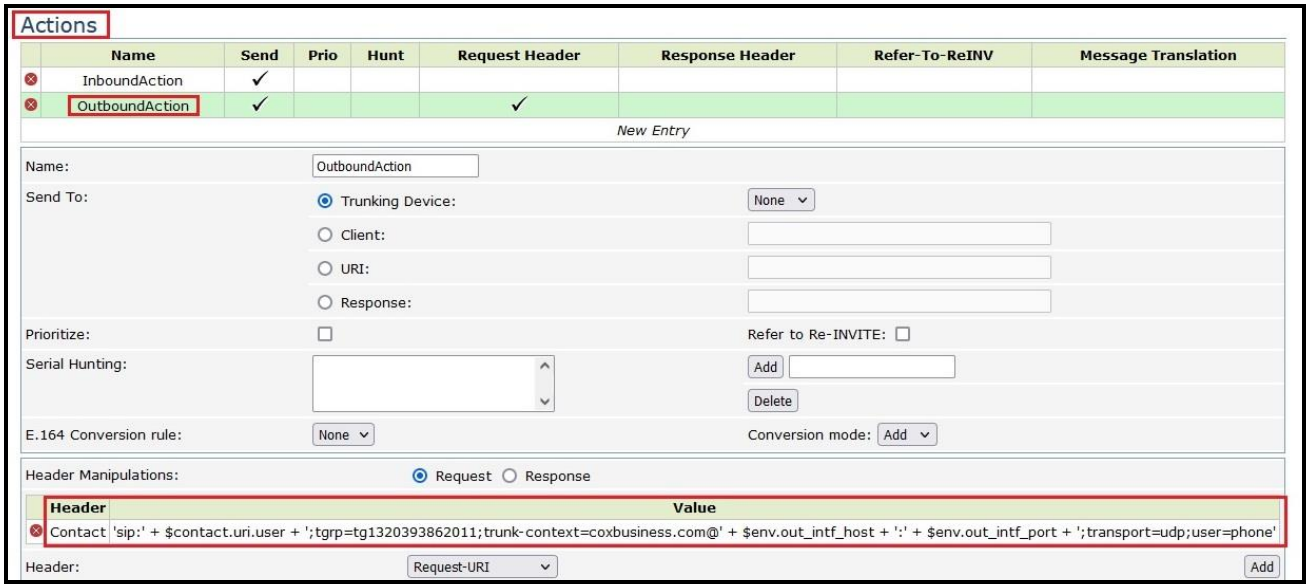


Figure 75 – Cox Managed CPE Outbound Action Configuration

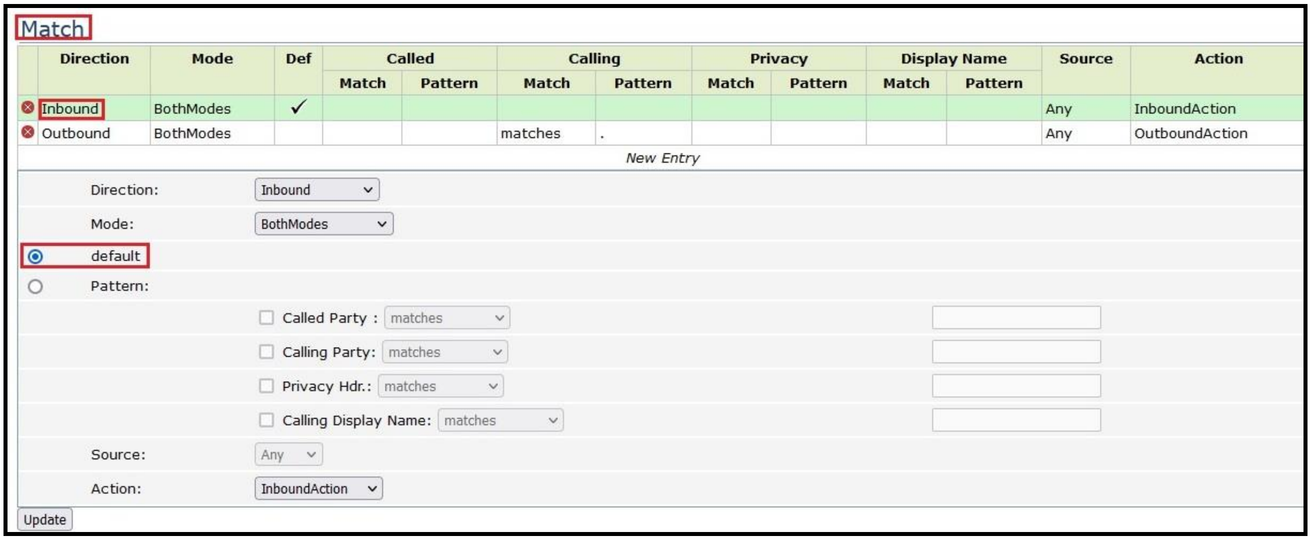


Figure 76 – Cox Managed CPE Inbound Match Configuration

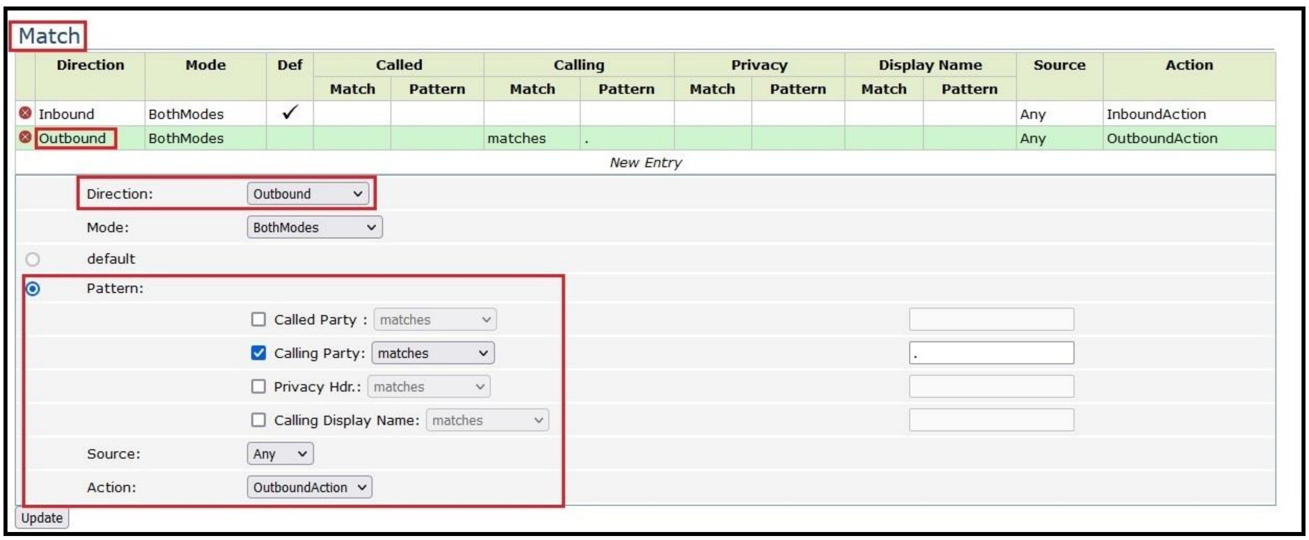


Figure 77 – Cox Managed CPE Outbound Match Configuration

11.6. Trunking Group Configuration

From the **Configuration Menu**, select **VoIP** menu option → **SIP** → **Trunking Group**.
Under **Create New Routing Group**

- Input a recognizable **Name** for the trunking Group: **Avaya**
- Input IP Address of **10.33.10.59**
- Click on **Create** and **Save**

Figure 78 – Cox Managed CPE Trunking Group

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.

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