Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Wesley Clover Solutions Trading Platform with Avaya IP Office using SIP Trunks – Issue 1.0

Abstract

These Application Notes contain interoperability instructions for configuring Wesley Clover Solutions Trading Platform with Avaya IP Office. Compliance testing was conducted to verify the interoperability.

Testing was performed using Avaya IP Office 500 V2 R8.1, but it also applies to Avaya IP Office Server Edition R8.1 (single site configuration only).

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

Wesley Clover Solutions Trading Platform consists of an IP PBX and IP Turrets. Wesley Clover Solutions IP PBX communicates to Avaya IP Office via a SIP trunk using the UDP protocol. Wesley Clover Solutions IP turrets register with Wesley Clover Solutions IP PBX.

2. General Test Approach and Test Results

The compliance test focused on the interoperability between Avaya IP Office and Wesley Clover Solutions IP PBX.

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.

2.1 Interoperability Compliance Testing

Compliance testing focused on verifying call scenarios mentioned below:

- Call setup and termination
- Codec Negotiation for G.711MU, G.711A and G.729
- DTMF transmission using RFC2833
- Call Hold, Call Transfers and Conference calls

2.2 Test Results

All executed test cases were passed and all objectives were met with the observation noted below:

- For call scenarios related to Call Conferences, Transfers and Call Forwards, Wesley Clover Solutions IP PBX holds onto the SIP trunk member for each call leg.

2.3 Support

Support for Wesley Clover Solutions can be found at:
Web: www.wesleycloversolutions.com
E-mail: service@wesleycloversolutions.com

3 Reference Configuration

The following figure displays the configuration used during the compliance test. The configuration below displays Wesley Clover Solutions IP PBX connected to Avaya IP Office 500 V2 using a SIP trunk. Endpoints for Wesley Clover Solutions IP PBX and Avaya IP Office are connected to a switch on the same network.

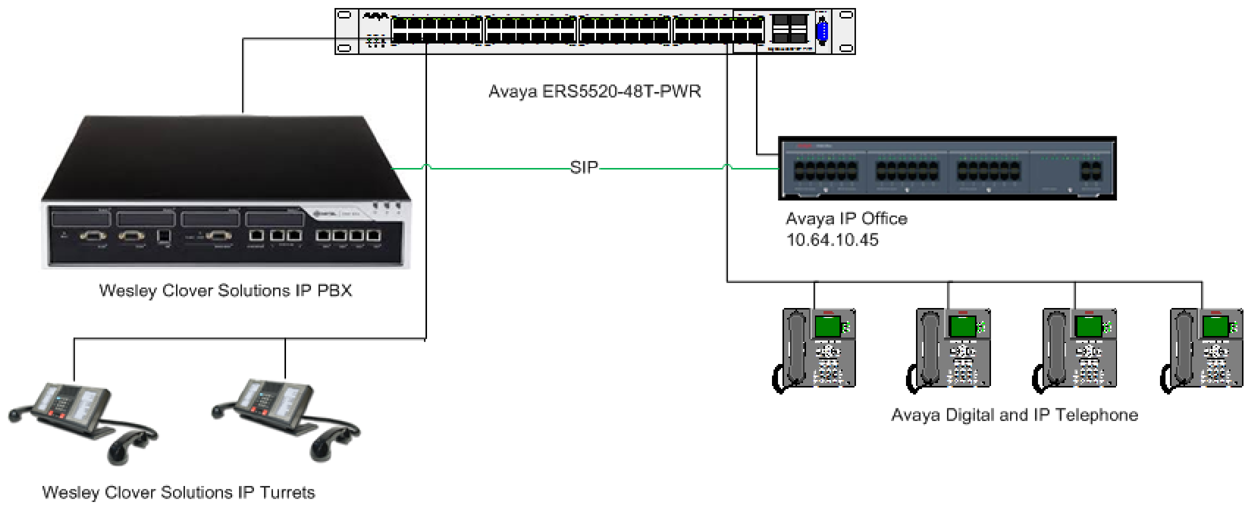


Figure 1: Reference Configuration for Wesley Clover Solutions IP PBX

4 Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office 500 V2	8.1(69)
Avaya 96xx Series Deskphones – H.323	3.22
Avaya 12x0 Series Phones – SIP	4.3.18
Wesley Clover Solutions IP PBX	12.0.1.24
Wesley Clover Solutions IP Turrets	3.0.0.8

5 Configure Avaya IP Office

The configuration of Avaya IP Office system was performed using the Avaya IP Office Manager (from here on referred as Manager) application. Once completed, the Avaya IP Office Manager Configuration must be saved and uploaded to the IP Office System. This process may sometimes force a system reboot.

The Avaya IP Office configuration includes following sections:

- Connect to IP Office using Manager
- Verify IP Office Control Unit
- Configure System Parameters
- Verify IP Office Licenses
- Configure IP Office SIP Line
- Configure Short Codes
    - Routing to Wesley Clover IP PBX
- Configure Incoming Call Routes
    - Calls Received on the SIP Line
- Saving IP Office Configuration

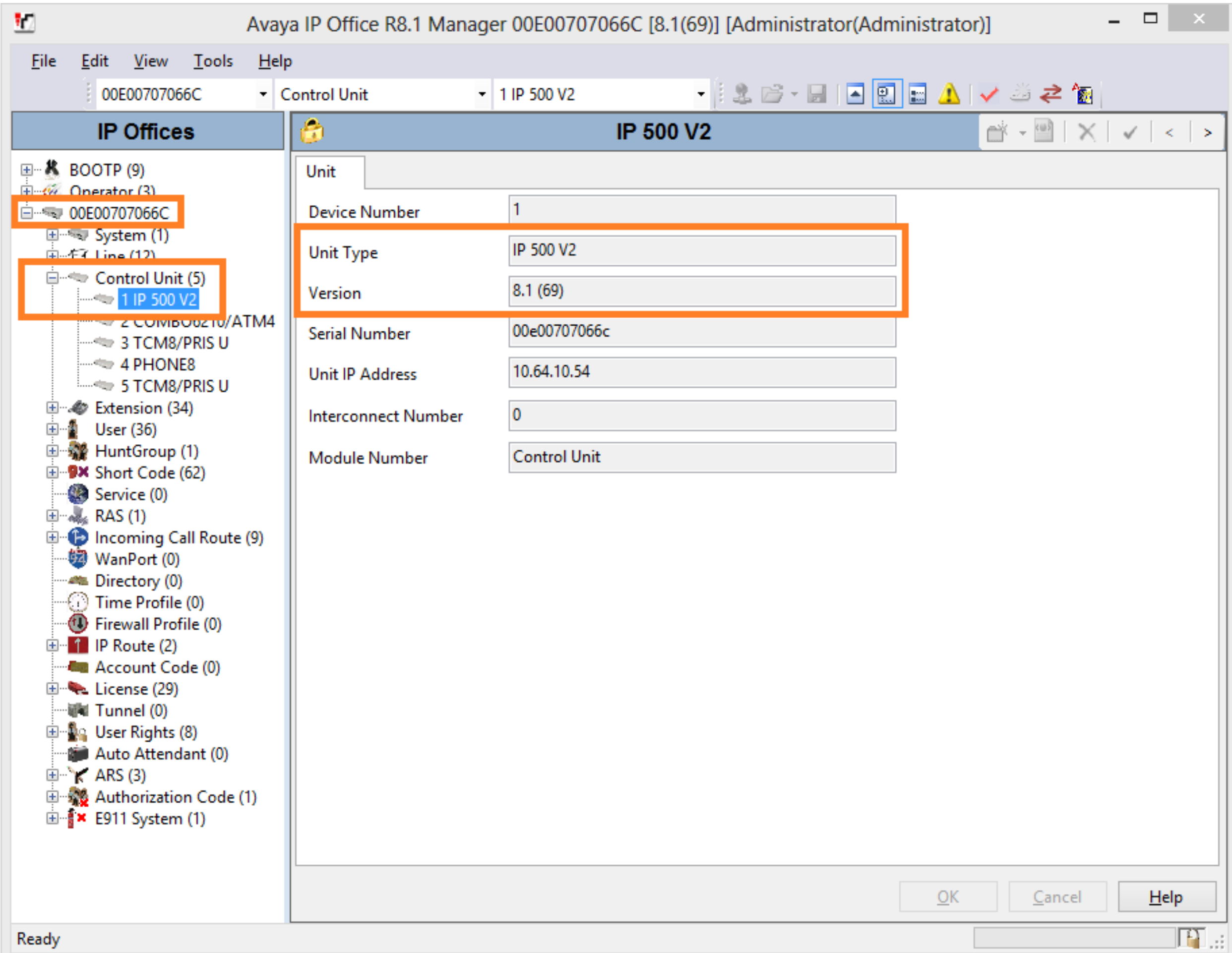
5.1 Connect to Avaya IP Office using Avaya IP Office Manager

From a Windows PC, open **Manager**. Location of the **Manager** will vary depending on the Windows Operating System. For Windows XP, navigate to **Start → All Programs → IP Office → Manager**. In the IP Offices window expand the Configuration Tree and double-click **System**. For this compliance test the IP Office System was called 00E00707066C. All configuration is performed under this system.

5.2 Verify Avaya IP Office Control Unit

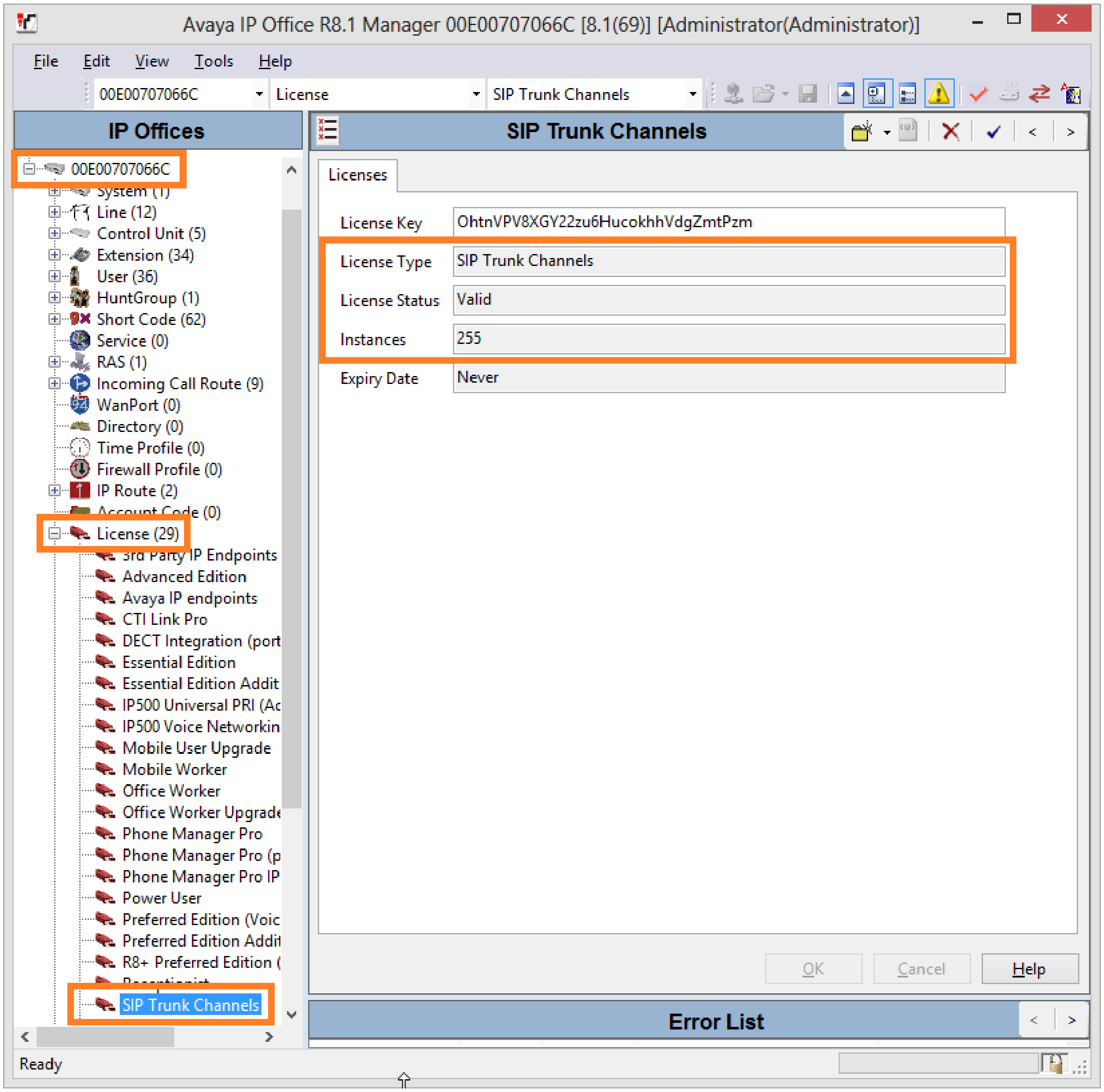
From the configuration tree in the navigation pane on the left, navigate to **IP 500 V2** as shown in the screen shot below.

Verify **Unit Type** and **Version**. During compliance test, Avaya IP Office 500 V2 hardware with version 8.1 software was used.



5.3 Verify Avaya IP Office Licenses

From the left pane, expand **License** and highlight **SIP Trunk Channels**, as shown in the screen capture. Verify the **License Status** field displays **Valid**.



5.4 Configure System Parameters

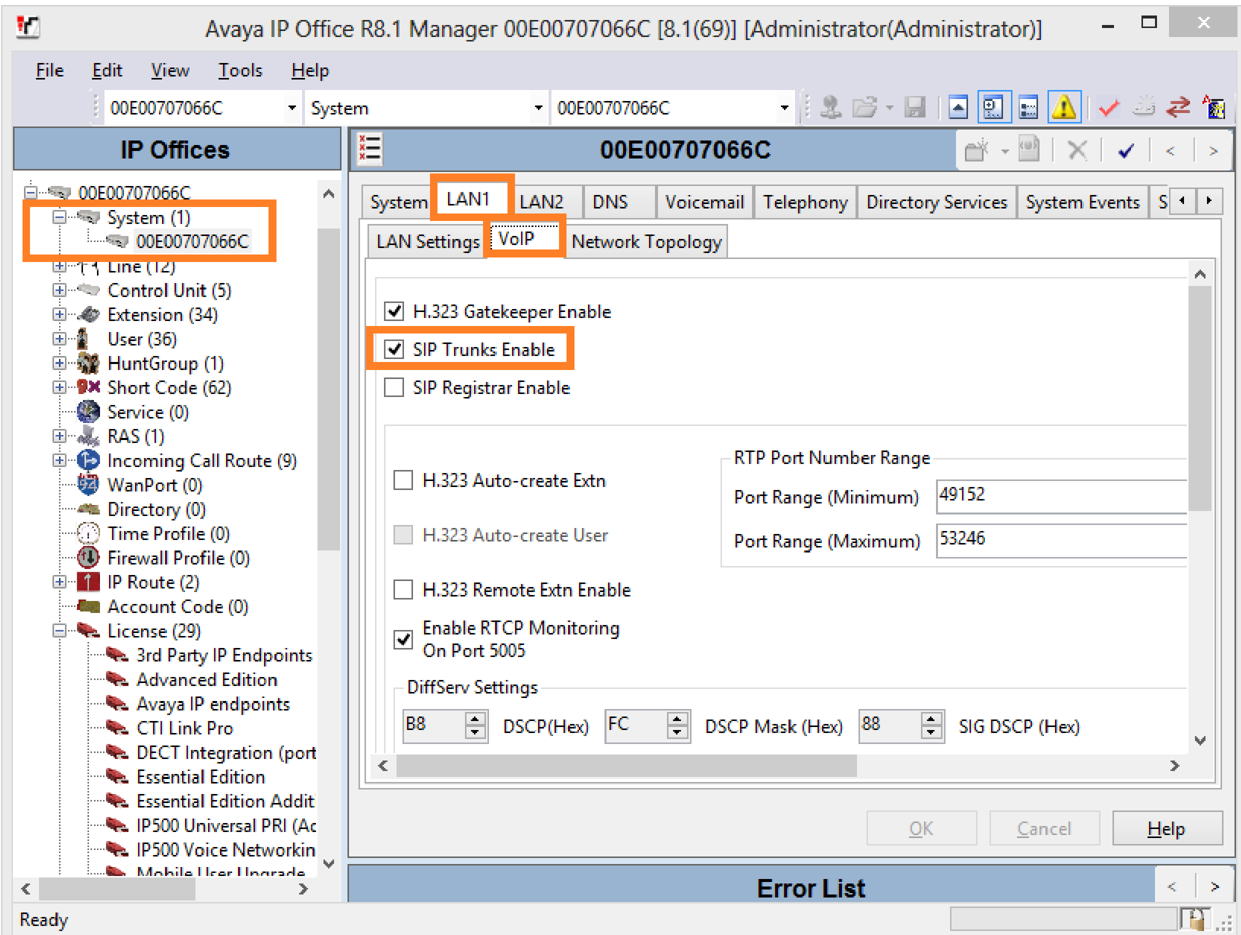
From the left pane, expand **System** then highlight **LAN1** in the right pane. From this page the **LAN Settings** and **VoIP** tabs will need to be configured.

5.4.1 LAN Settings

It is assumed that LAN Settings have already been configured.

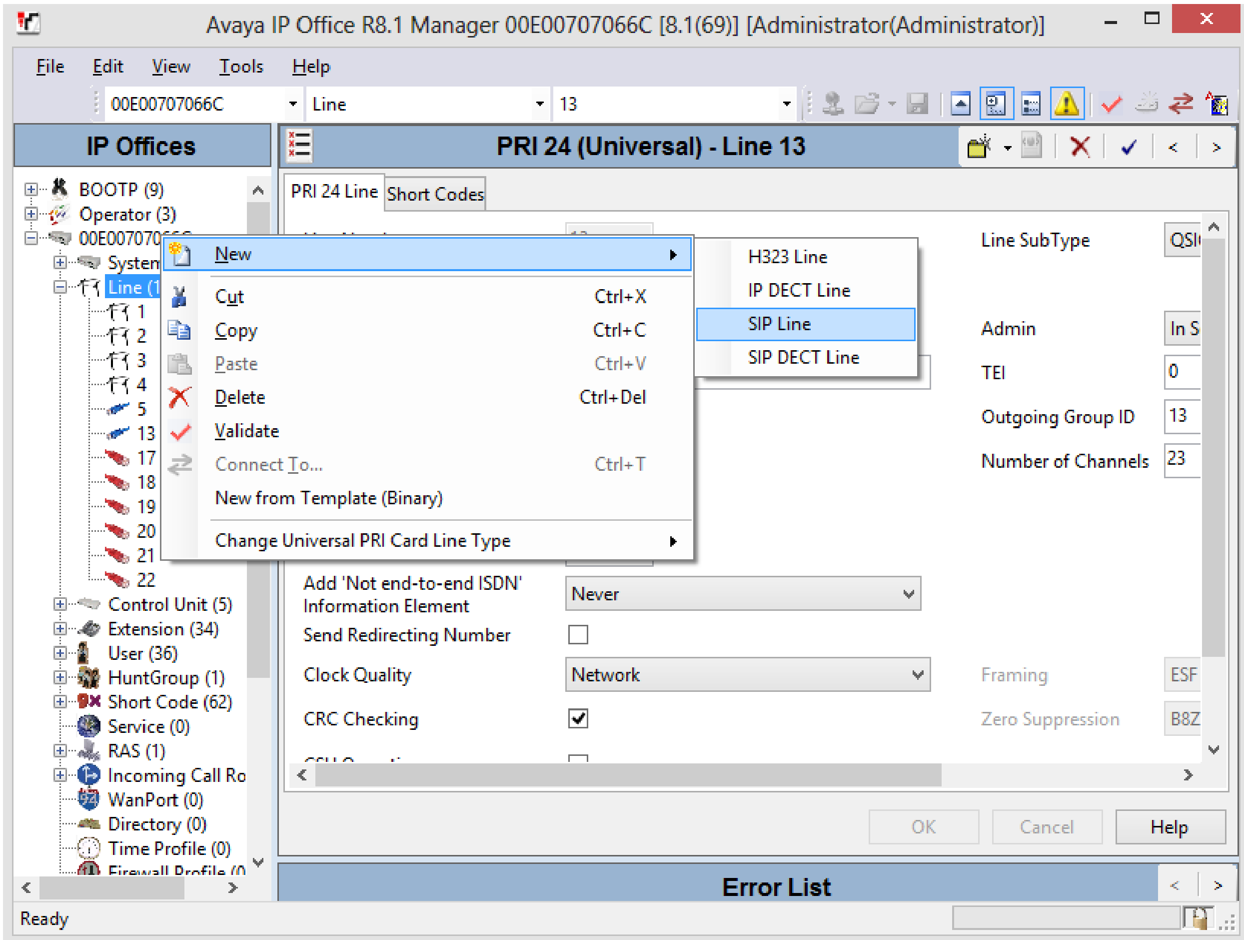
5.4.2 VoIP

From the **VoIP** tab. Verify that the **SIP Trunks Enable** box is checked.

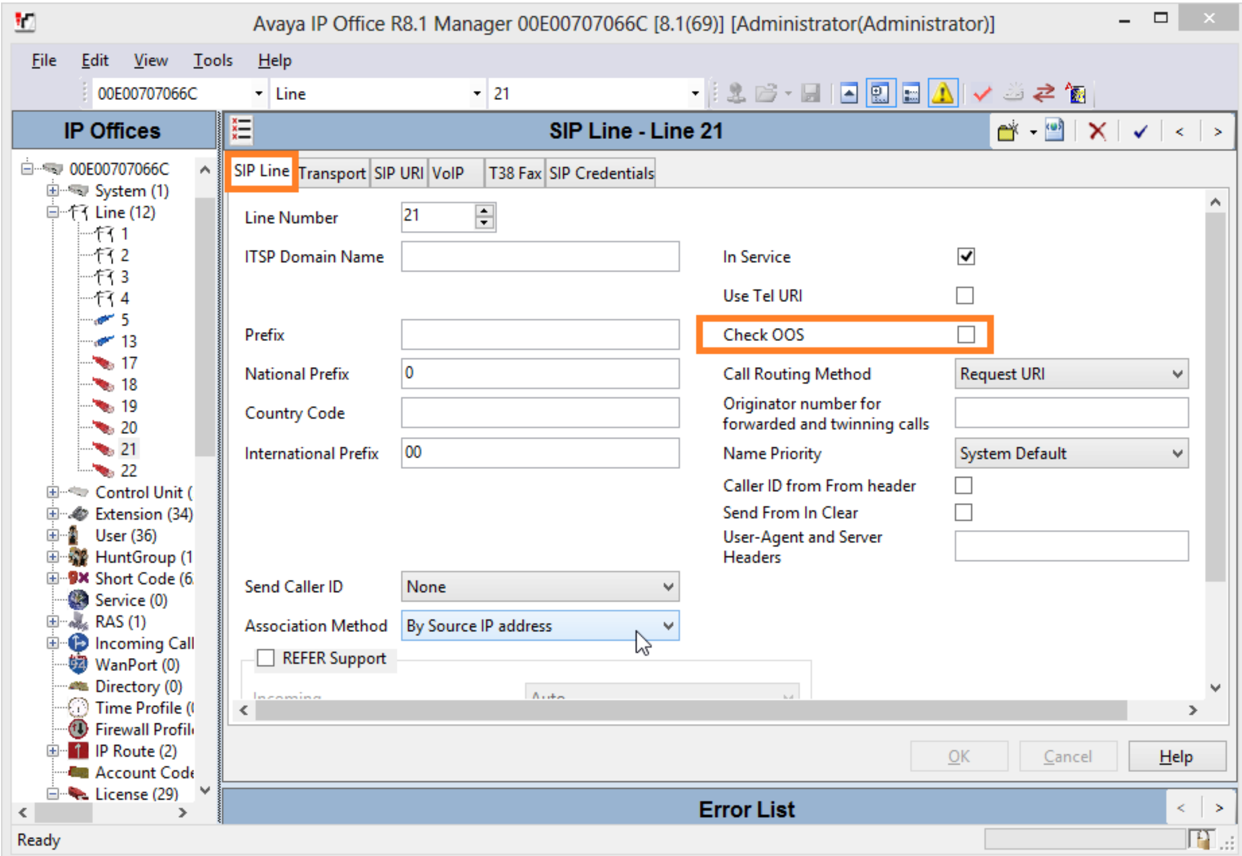


5.5 Configure Avaya IP Office SIP Line

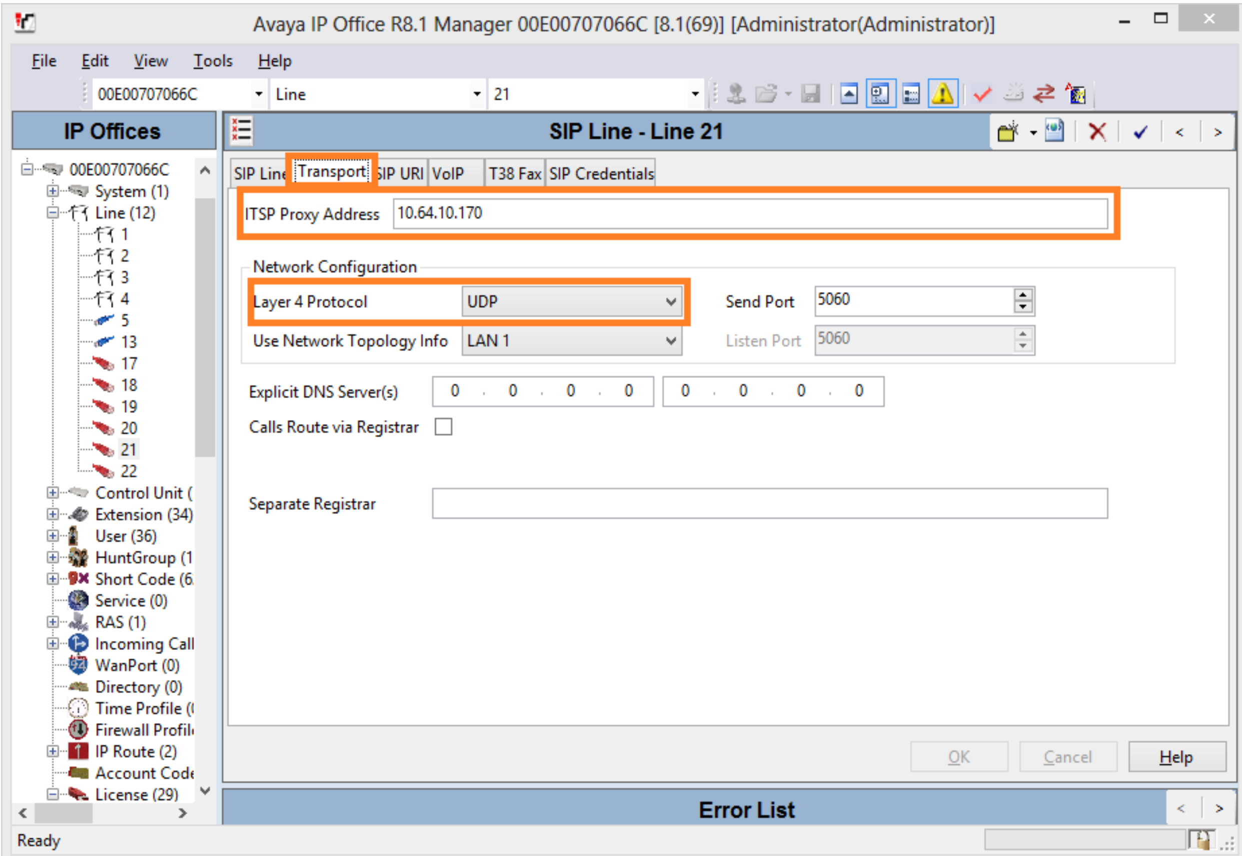
On the left lane, right click on **Line**, select **New → SIP Line**.



Under **SIP Line** tab, uncheck box for **Check OSS**.

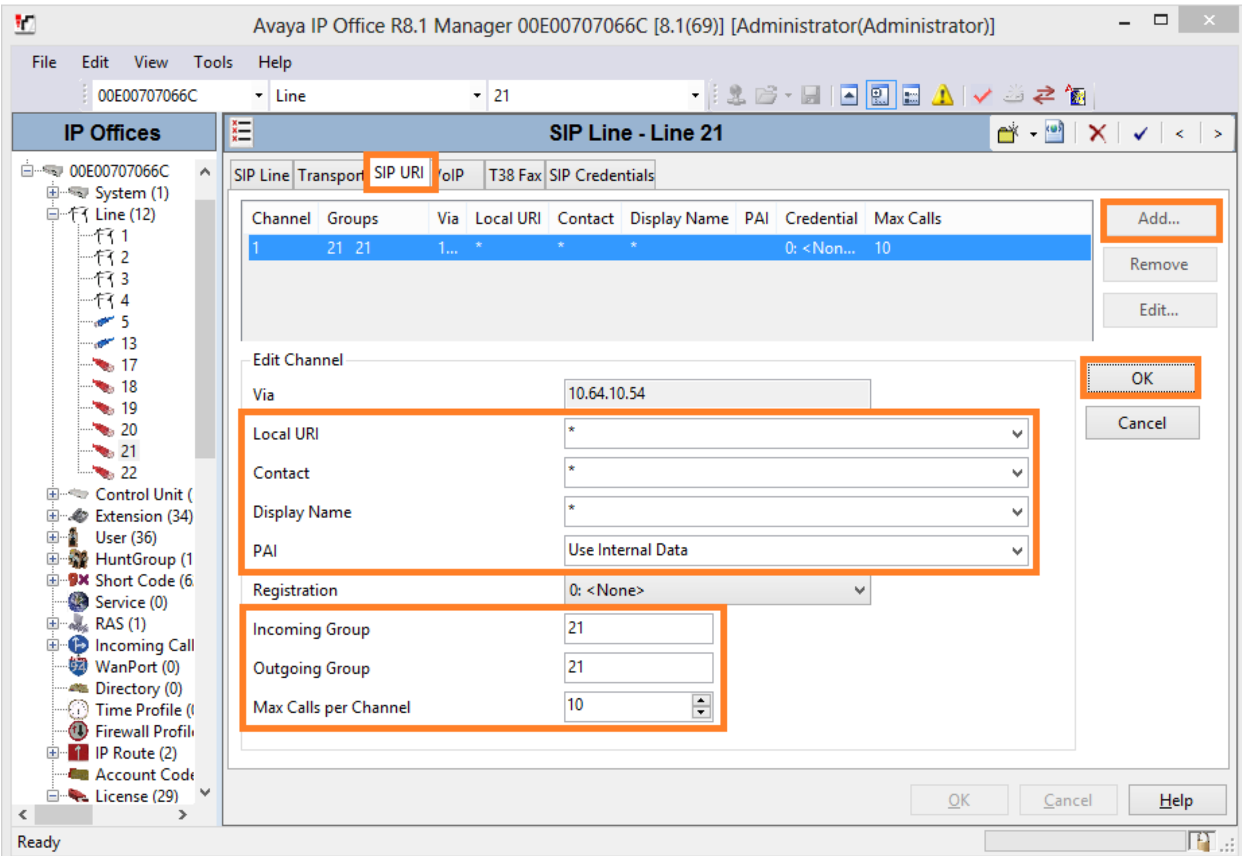


Under the **Transsport** tab, in **ITSP Proxy Address**, type in the IP address of Wesley Clover Solutions IP PBX. Set **Layer 4 Protocol** to **UDP**.



Under **SIP URI**, click the **Add…** button on the right side to add SIP Channels to Wesley Clover Solutions IP PBX.

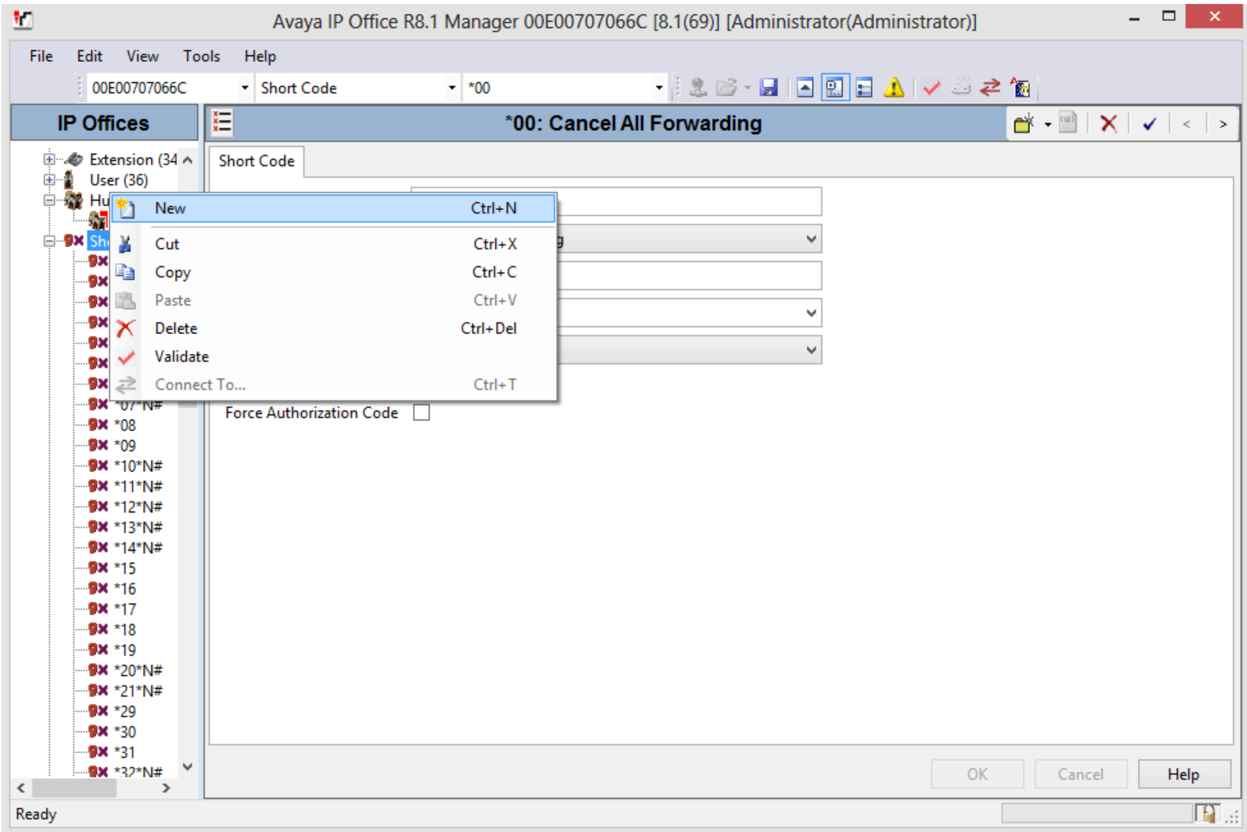
- Set **Local URI, Contact** and **Display Name** to *****.
- In **Incoming Group** and **Outgoing Group**, type in the SIP Line number that was configured in **Section 5.5**.
- Set the number of **Max Calls per Channel**. During compliance testing, 10 **Max Calls per Channel** was used.



At the bottom of the window, click **OK**, to save the configuration for SIP Line.

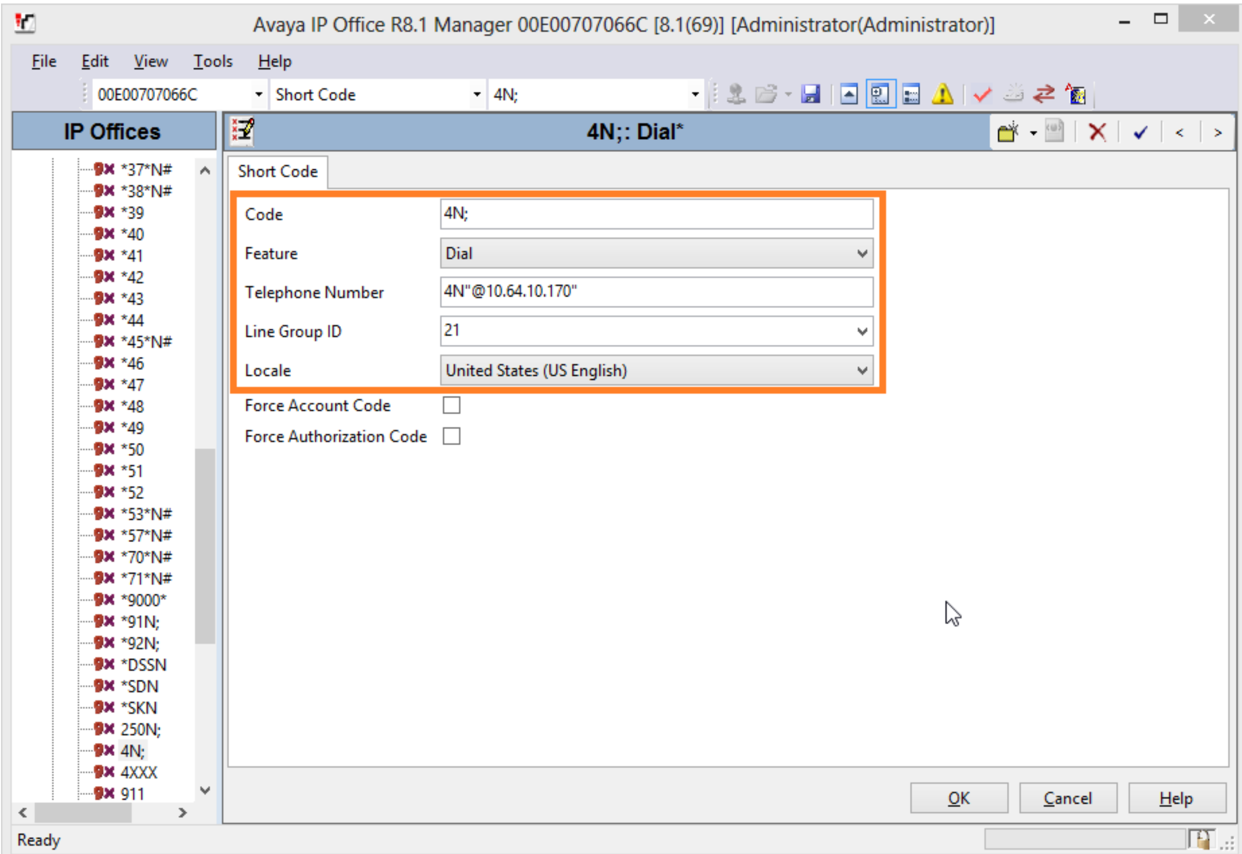
5.6 Configure Short Code for routing to Wesley Clover Solutions IP PBX

From the left pane, right click on **Short Code** and select **New**.



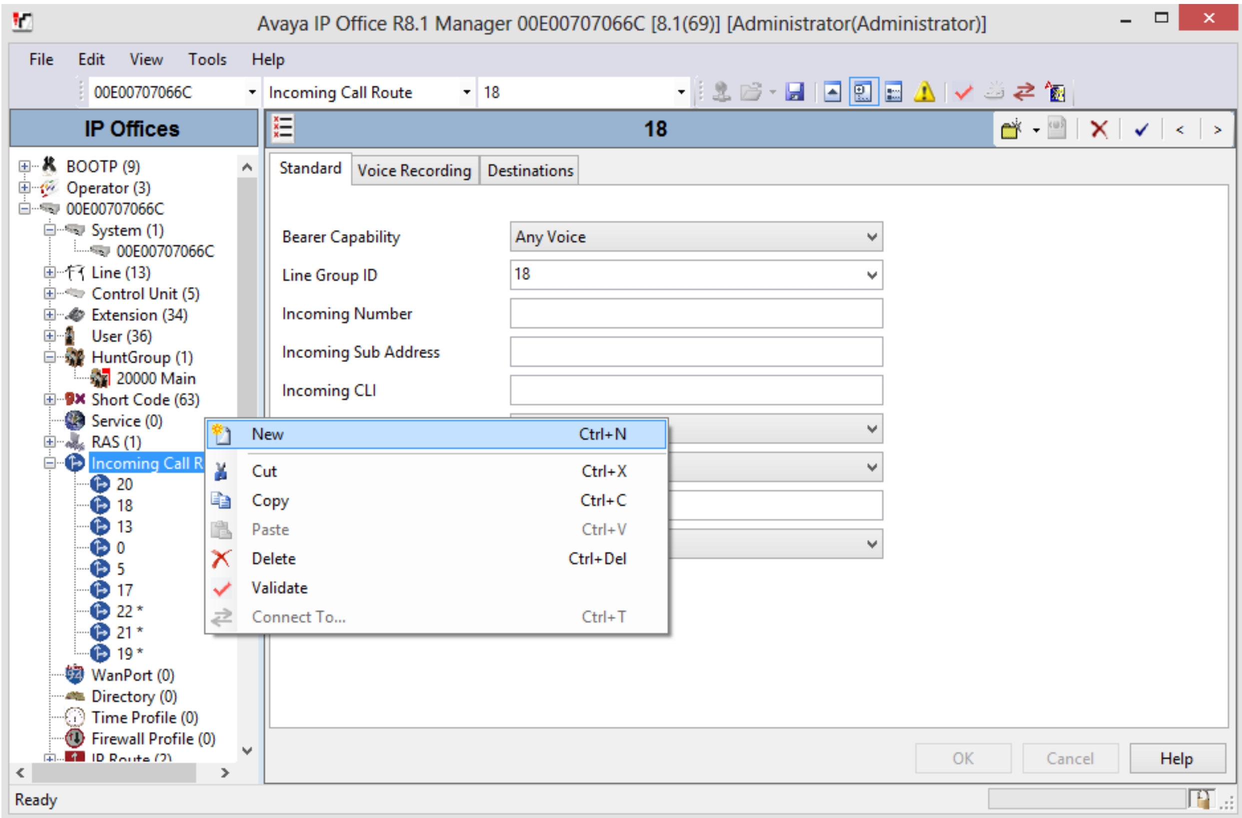
The configured short code will route calls to Wesley Clover Solutions IP PBX when any number starting with 4 is dialed:

- Type in **4N;** in **Code** field.
- Set **Feature** to **Dial.**
- Type in **4N"@ip-address"** for **Telephone Number**, where ip-address is the IP Address of Wesley Clover Solutions IP PBX.
- Type in the number of PRI Line that was configured **Section 5.6**, in **Line Group ID** field.



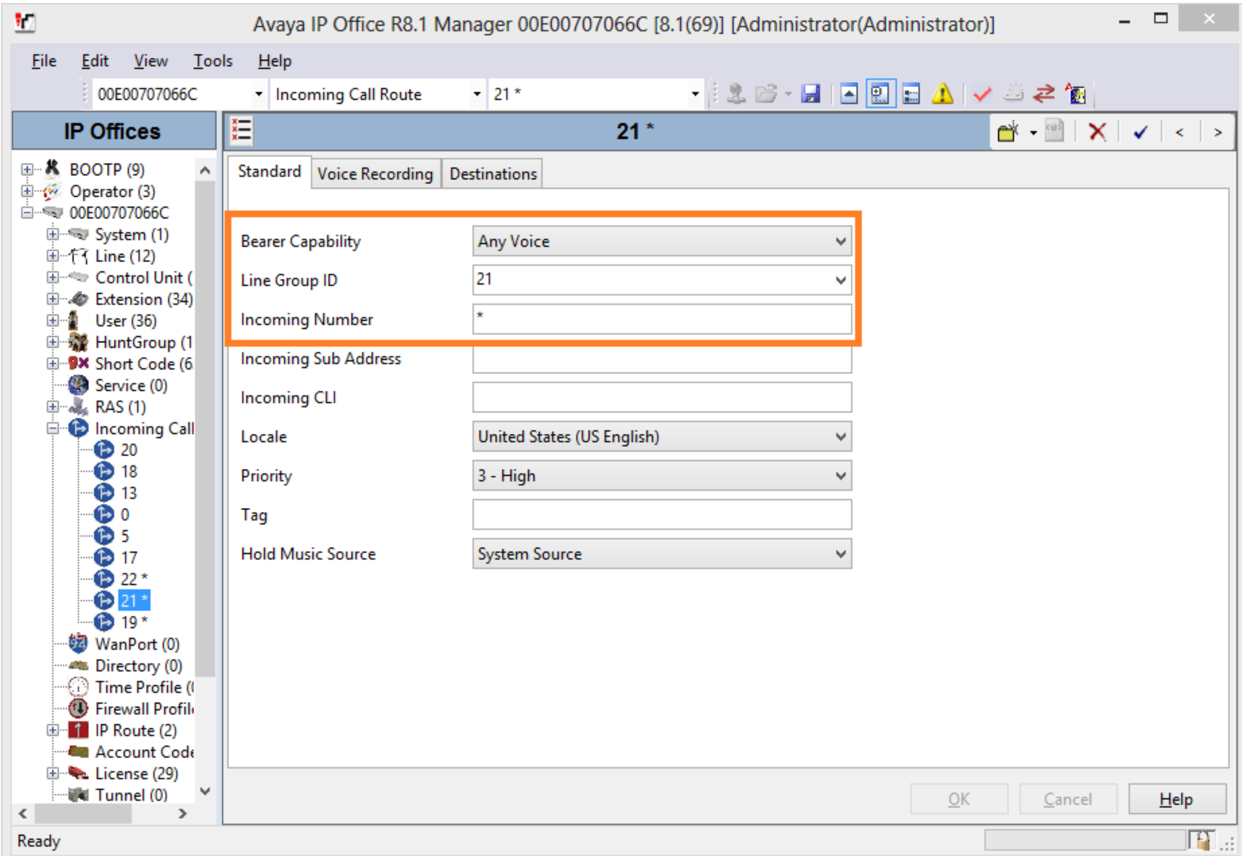
5.7 Configure Incoming Call Routes

From the left pane, right click on **Incoming Call Route,** and select **New**.



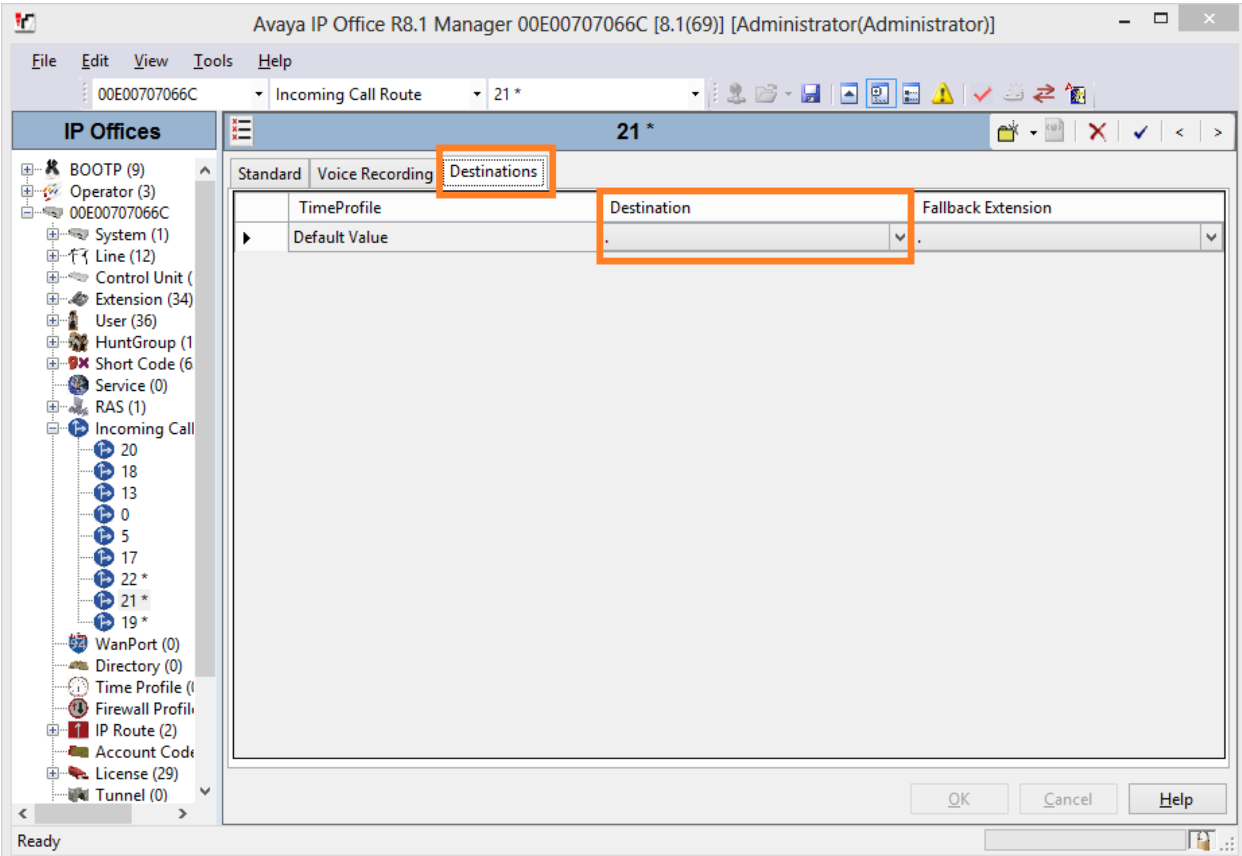
Under the **Standard** tab, configure the following:

- Set **Bearer Capability** to **Any Voice**
- Set **Line Group ID** to Line Group ID of the SIP Line defined earlier (21)
- Set **Incoming Number** to *



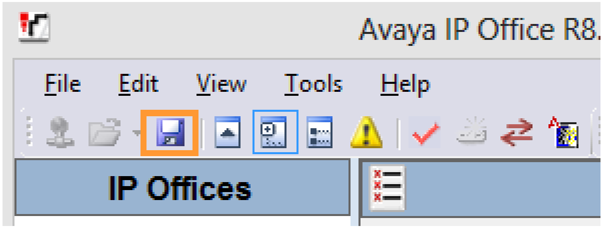
Under the **Destinations** tab, configured the following:

- Under **Destination** column, type in a dot, "**.**" This will allow to route incoming call based on the number that was dialed.



5.8 Saving IP Office Configuration

Once the configuration changes have been completed, select the floppy disk icon to push the changes to the IP Office system.



Note: *Changes will not take effect until this step is completed. This may cause a reboot of Avaya IP Office causing service disruption.*

6 Configure Wesley Clover Solutions

Wesley Clover Solutions trading platform utilizes Wesley Clover Solutions IP PBX, to allow for call routing via SIP trunks for inter-pbx and external call routing. The following information provides programming guidelines for the SIP connection between Wesley Clover Solutions IP PBX and Avaya IP Office.

6.1 Assumptions

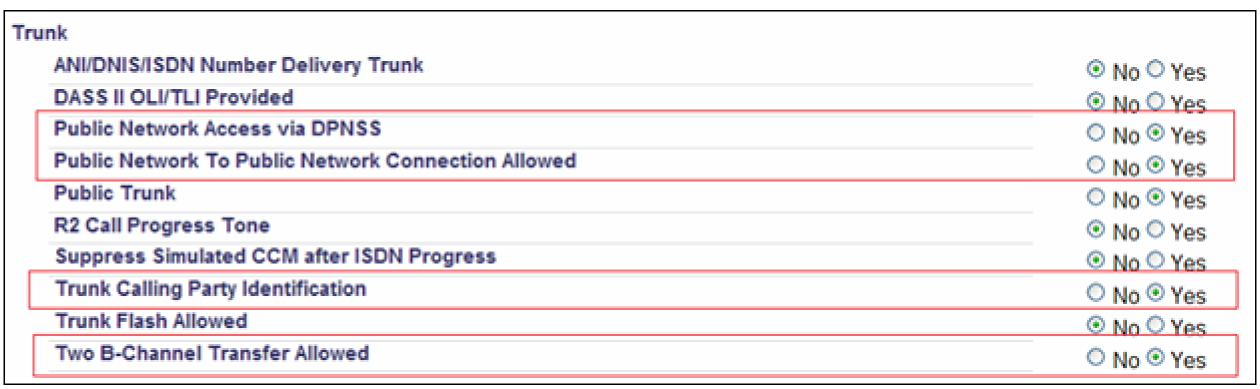
- It is assumed for the purposes of this document that the appropriate number of SIP trunk licenses have been applied in Wesley Clover Solutions IP PBX.
- The dial-able Avaya extension numbers are 4 digits in length.
- There are no dial restrictions to Avaya IP Office.

Note: Configuration is performed via a web browser, by navigating to http://<**ip-address**> , where **ip-address** is the IP address of Wesley Clover Solutions IP PBX.

6.2 Program Class of Service

Navigate to **System Properties→System Feature Settings→Class of Service Options** (not shown).
Program a unique COS, in this case **6** is used (not shown) and set the following trunk options to **Yes**:
Verify that the following options are set to "Yes"

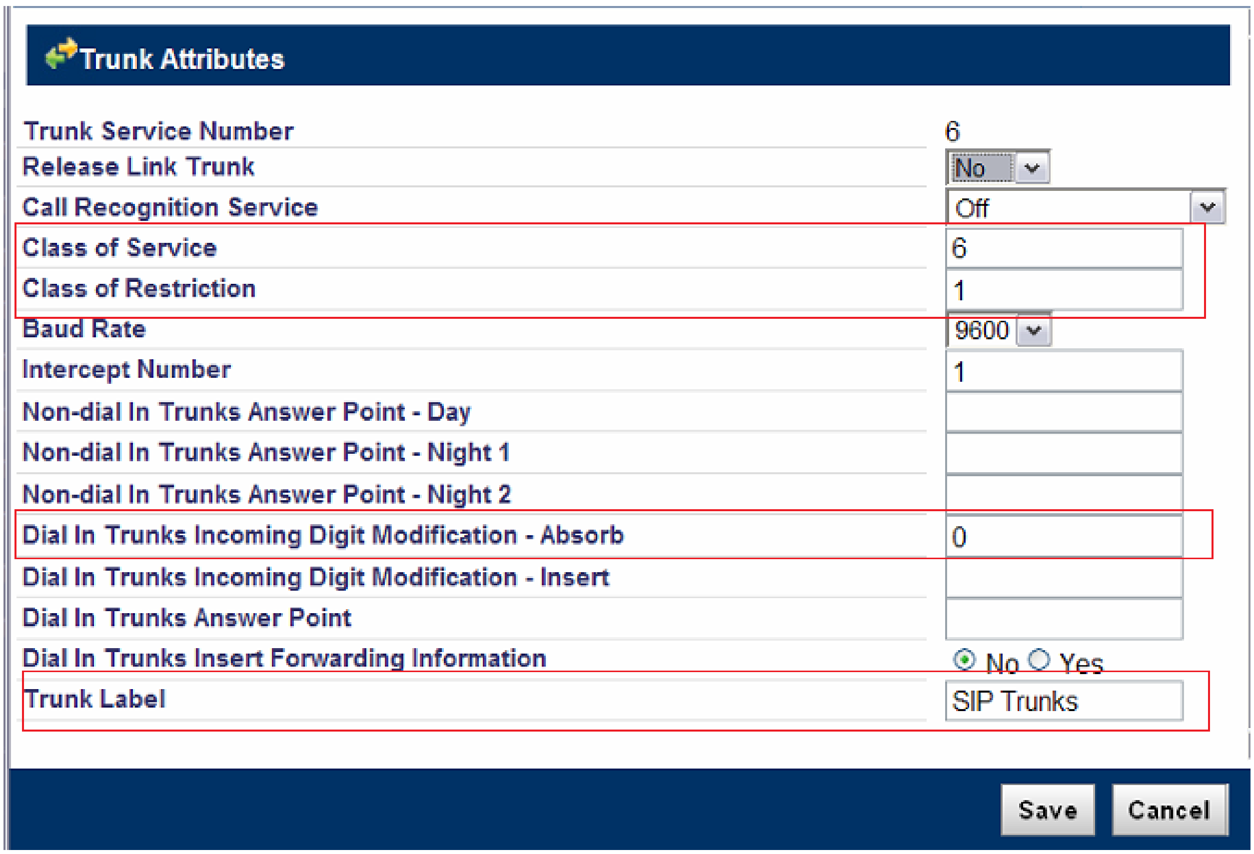
- Public Network Access via DPNSS
- Public Network To Public Network Connection Allowed
- Trunk Calling Party Identification
- Two B-Channel Transfer Allowed



6.3 Program Trunk Attributes Form

Navigate to **Trunks→Trunk Attributes** (not shown)
In this example, **6** is used as a Trunk Service Number.

- Set the **Class of Service** to the COS assigned in **Section 6.2**.
- Set **Class of Restriction** to **1**.
- Set the **Dial-In Trunk Incoming Digit Modification – Absorb** to **0**.
- Add a **Trunk Label**.



Trunk Attributes

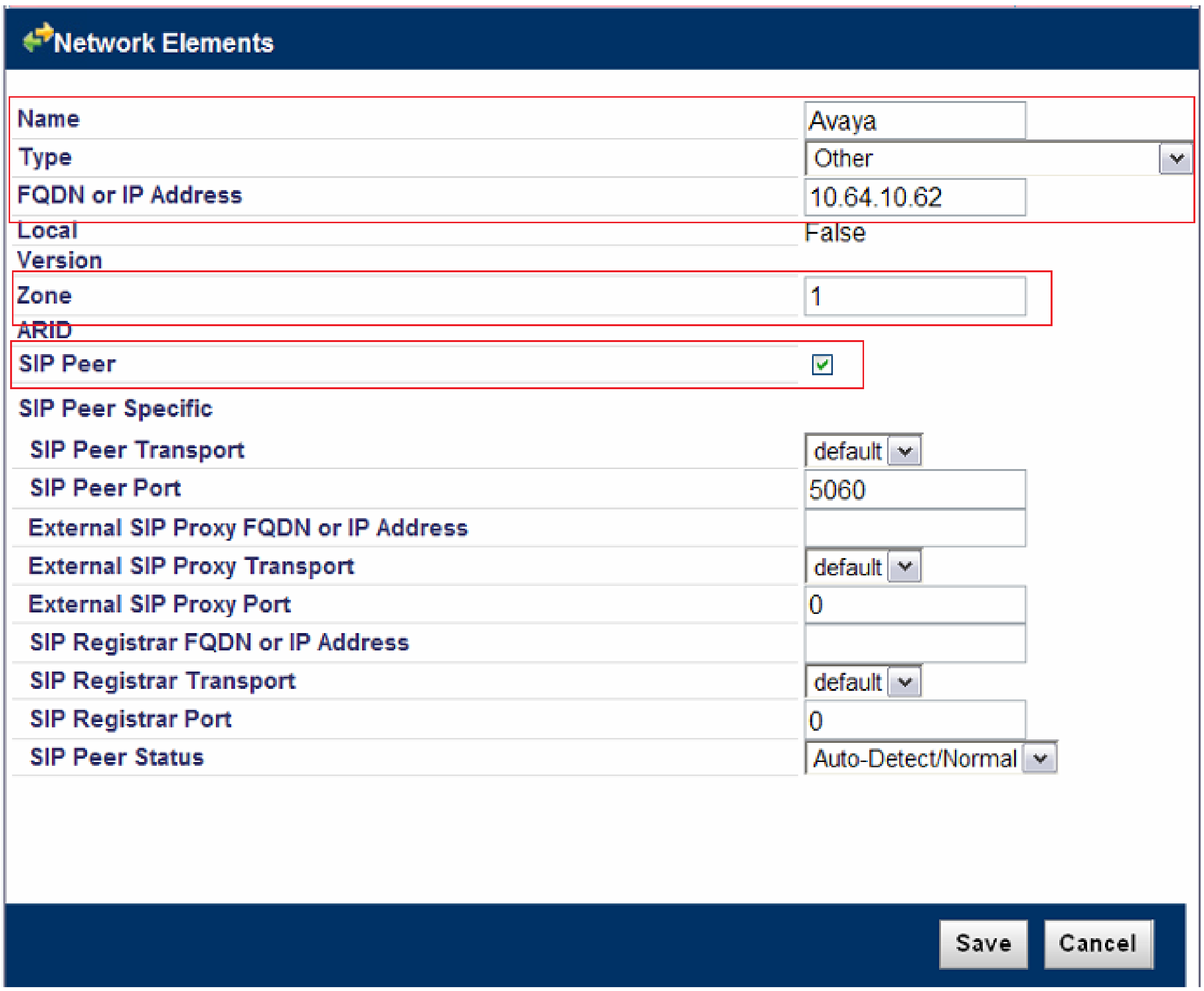
Trunk Service Number	6
Release Link Trunk	No
Call Recognition Service	Off
Class of Service	6
Class of Restriction	1
Baud Rate	9600
Intercept Number	1
Non-dial In Trunks Answer Point - Day	
Non-dial In Trunks Answer Point - Night 1	
Non-dial In Trunks Answer Point - Night 2	
Dial In Trunks Incoming Digit Modification - Absorb	0
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	No Yes
Trunk Label	SIP Trunks

Save Cancel

6.4 Program the Network Elements Form

Navigate to **Voice Network → Network Elements** (not shown)
Configure the network element as follows:

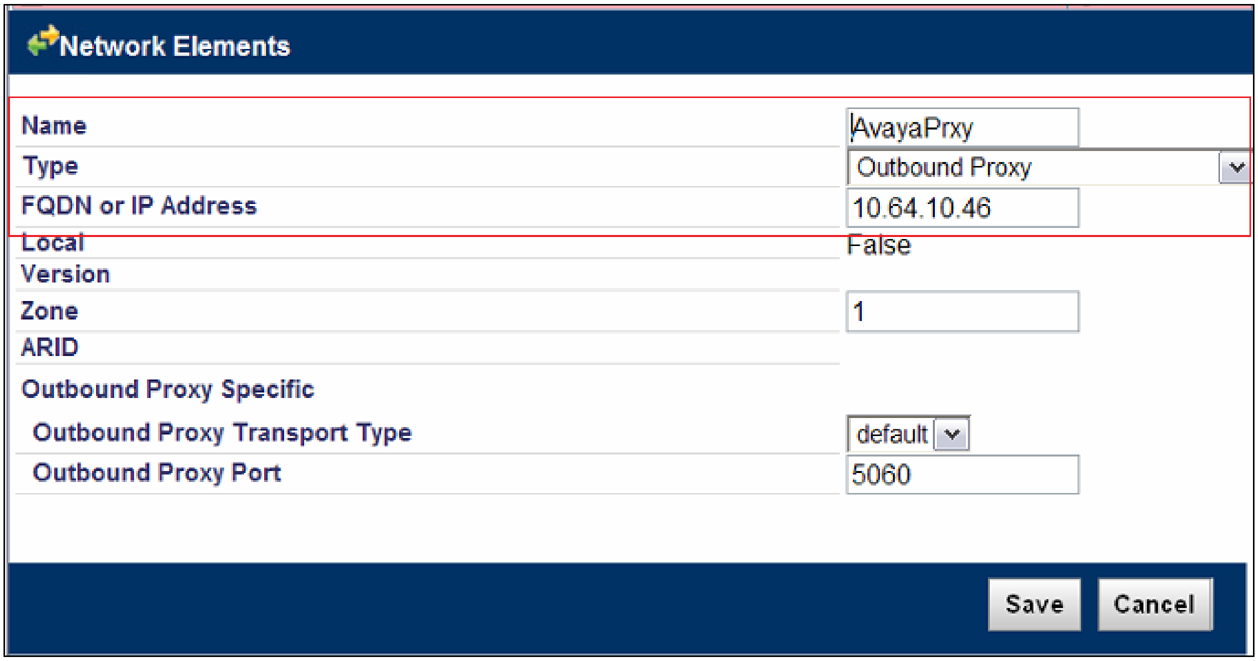
- Enter a name in the **Name** Field. For example, "Avaya".
- Select **Other** in the **Type** drop down box.
- Enter the IP address of IP Office in **FQDN or IP Address** field.
- Set **Zone** of 1.
- Select **SIP Peer** selection box.



Program the Network Elements Form (Continued)

Configure a second Network Element for the proxy.

- Enter a meaningful name in the **Name** field. For example, “AvayaPrxy”.
- Select **Outbound Proxy** in the **Type** drop down.
- Enter the IP address of IP Office in the **FQDN or IP Address** field.

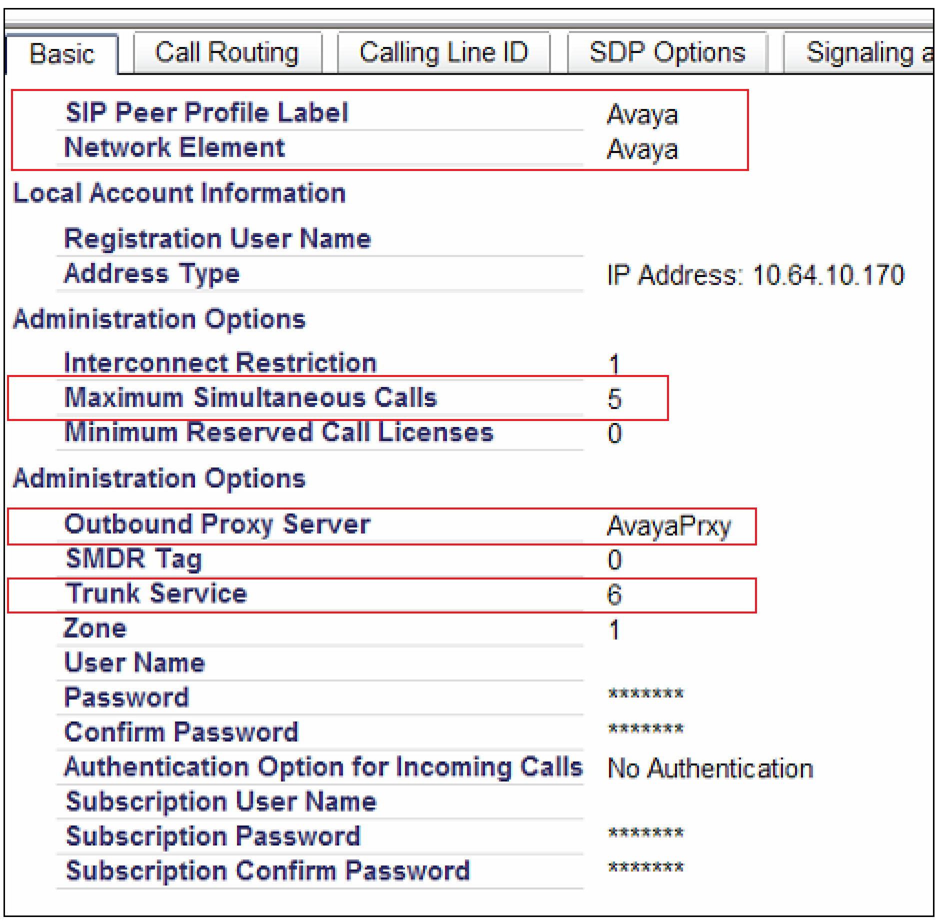


6.5 Program SIP Peer Profile Form

Navigate to **Trunks → SIP → SIP Peer Profile** (not shown).

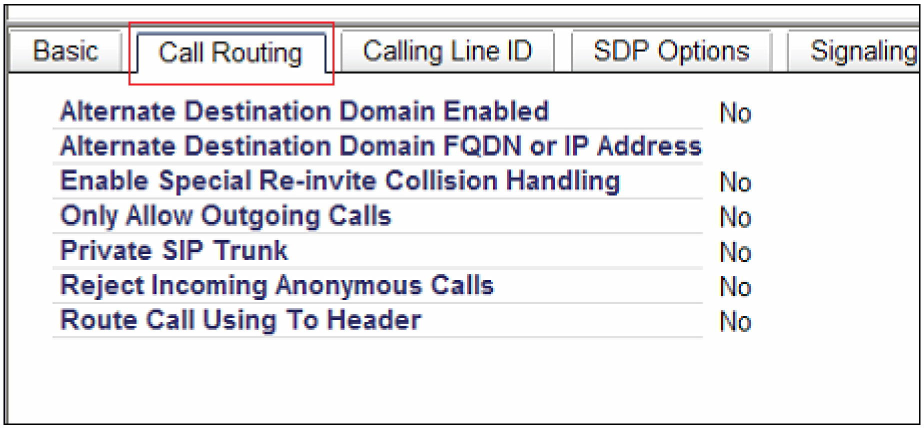
In the **SIP Peer Profile**, add a new peer based on the following screen capture. Click the Add button to begin creating the new SIP Peer Profile (not shown). Under the **Basic** tab, configure as follows:

- Type in a meaningful name in **SIP Peer Profile Label** field. For example, "Avaya".
- In the Network Element drop down box select the Network Element created in **Section 6.4**.
- Leave the **Registration User Name** field blank.
- Enter the **Maximum Simultaneous Calls**. This is the number of SIP trunks to be used between IP Office and Wesley Clover Solutions IP PBX. This number must be less than or equal to the number of SIP Trunk licenses applied to Wesley Clover Solutions IP PBX.
- In the **Outbound Proxy Server** drop down box select "AvayaPrxy" created in **Section 6.4**.
- In the Trunk Service field enter the **Trunk Service Number** created in **Section 6.3**.



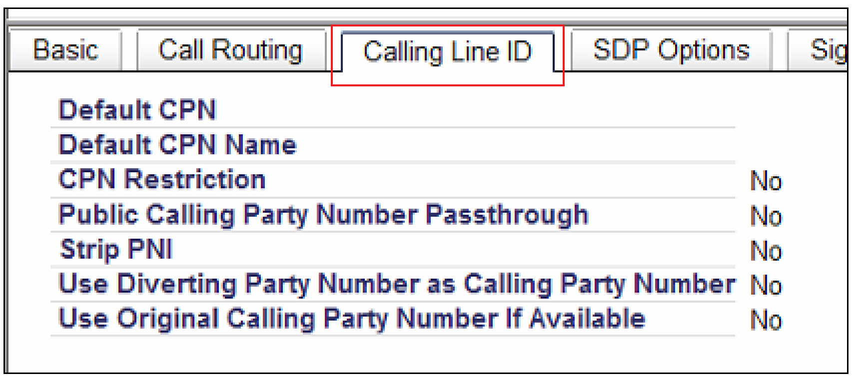
Program SIP Peer Profile Form (Continued)

Under each tab in the SIP Peer Profile Form, ensure all the options are configured as shown in the screen captures below:



Basic | Call Routing | Calling Line ID | SDP Options | Signaling

Alternate Destination Domain Enabled	No
Alternate Destination Domain FQDN or IP Address	
Enable Special Re-invite Collision Handling	No
Only Allow Outgoing Calls	No
Private SIP Trunk	No
Reject Incoming Anonymous Calls	No
Route Call Using To Header	No



Basic | Call Routing | Calling Line ID | SDP Options | Sig

Default CPN	
Default CPN Name	
CPN Restriction	No
Public Calling Party Number Passthrough	No
Strip PNI	No
Use Diverting Party Number as Calling Party Number	No
Use Original Calling Party Number If Available	No


Basic | Call Routing | Calling Line ID | SDP Options | Signaling

Allow Peer To Use Multiple Active M-Lines	No
Allow Using UPDATE For Early Media Renegotiation	No
Avoid Signaling Hold to the Peer	No
Enable Mitel Proprietary SDP	No
Force sending SDP in initial Invite message	Yes
Force sending SDP in initial Invite - Early Answer	No
Ignore SDP in Unreliable Provisional Responses	No
Limit to one Offer/Answer per INVITE	No
NAT Keepalive	No
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes
Renegotiate SDP To Enforce Symmetric Codec	No
Repeat SDP Answer If Duplicate Offer Is Received	No
RTP Packetization Rate Override	No
RTP Packetization Rate	20ms
Special handling of Offers in 2XX responses (INVITE)	No
Suppress Use of SDP Inactive Media Streams	No

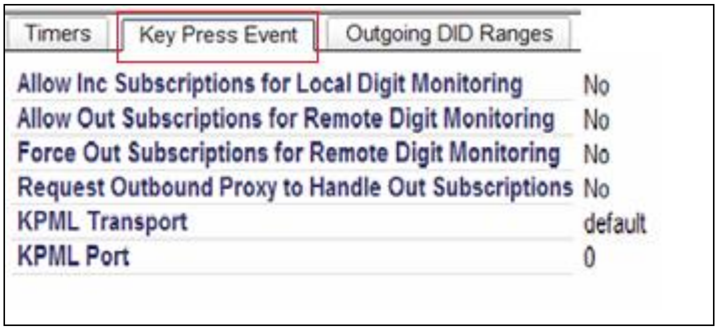
Program SIP Peer Profile Form (Continued)

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation

Trunk Group Label	
Allow Display Update	No
Build Contact Using Request URI Address	No
De-register Using Contact Address not *	No
Disable Reliable Provisional Responses	Yes
Disable Use of User-Agent and Server Headers	No
E.164: Enable sending '+'	No
E.164: Add '+' if digit length > N digits	0
E.164: Do not add '+' to Emergency Called Party	No
E.164: Do not add '+' to Called Party	No
Force Max-Forward: 70 on Outgoing Calls	No
If TLS use 'sips:' Scheme	No
Ignore Incoming Loose Routing Indication	No
Only use SDP to decide 180 or 183	No
Prefer From Header for Caller ID	No
Require Reliable Provisional Responses on Outgoing Calls	No
Use Fixed Retry Time for 491	No
Use Privacy: none	No
Use P-Asserted Identity Header	No
Use P-Asserted Identity for Billing	No
Use P-Preferred Identity Header	No
Use Restricted Character Set For Authentication	No
Use To Address in From Header on Outgoing Calls	No
Use user=phone	No

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers

Keep-Alive (OPTIONS) Period	120
Registration Period	3600
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	90
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0



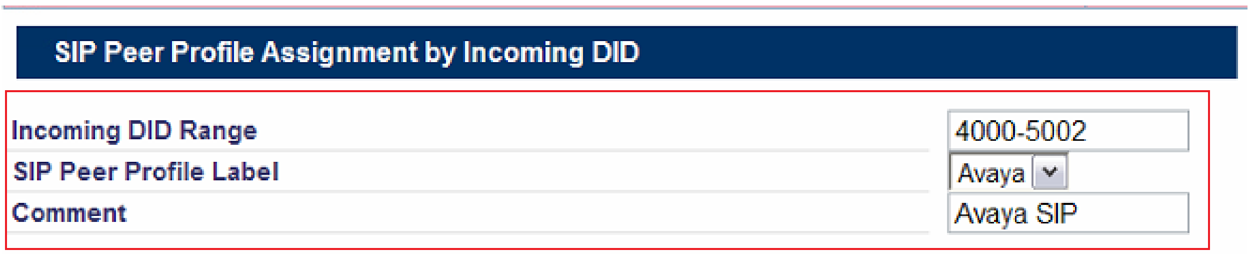
Timers | Key Press Event | Outgoing DID Ranges

Allow Inc Subscriptions for Local Digit Monitoring	No
Allow Out Subscriptions for Remote Digit Monitoring	No
Force Out Subscriptions for Remote Digit Monitoring	No
Request Outbound Proxy to Handle Out Subscriptions	No
KPML Transport	default
KPML Port	0

6.6 Program SIP Peer Profile Assignment by Incoming DID Form

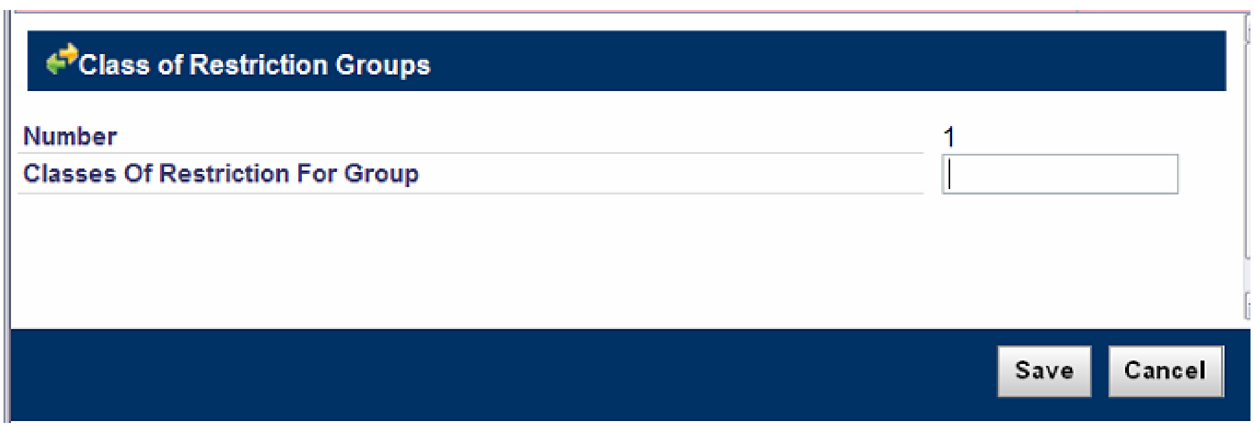
Navigate to **Trunks → SIP → SIP Peer Profile Assignment by Incoming DID** (not shown)
Add existing extension ranges to the Incoming DID Range. For the following, extension ranges of 4000-5002 are used:

- Click Add (not shown).
- Enter extension ranges in the **Incoming DID Range** field.
- Select the SIP Peer Profile Label created in **Section 6.5** in the drop down box.
- Add a meaningful comment in the Comment field.



6.7 Program the Class Of Restriction Group Form

Navigate to **System Properties→System Feature Settings→Class of Restriction Groups** (not shown)
Verify that the Class of Restriction has no restrictions. Choose an index number without any restrictions applied. In this example **Number 1** is used. Note that the **Classes of Restriction For Group** is blank indicating no restrictions.



6.8 Program ARS Digit Modification Plans Form

Navigate to **Call Routing→Automatic Route Selections (ARS) →ARS Digit Modification Plans**

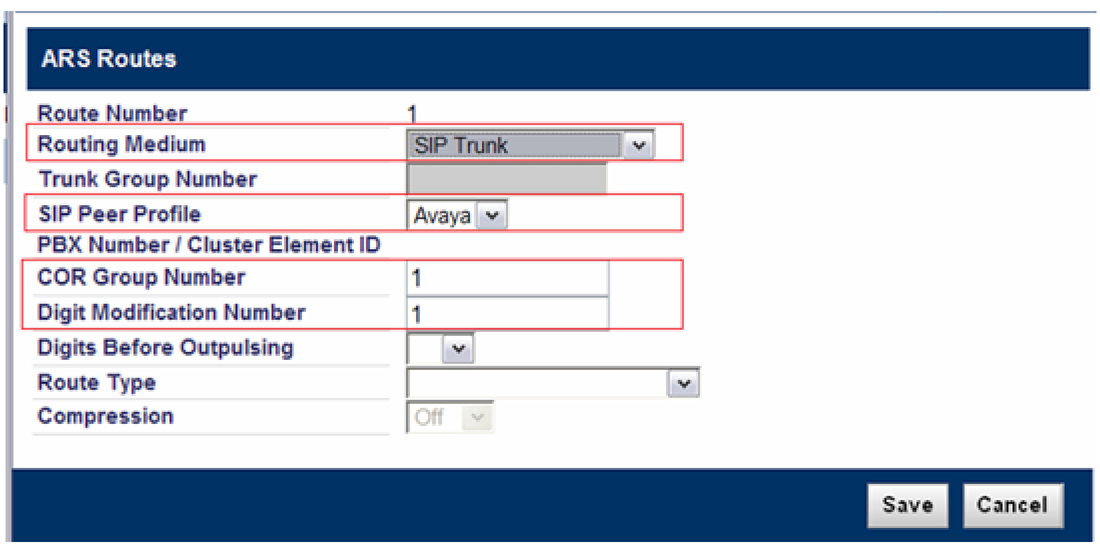
In this example **Digit Modification Number** of **1** is used. Set the **Number of Digits to Absorb** to **0**.

6.9 Program Route Assignment Form

Navigate to **Call Routing→Automatic Route Selection (ARS)→ARS Routes** (not shown)

In this example **Route Number** of **1** is used.

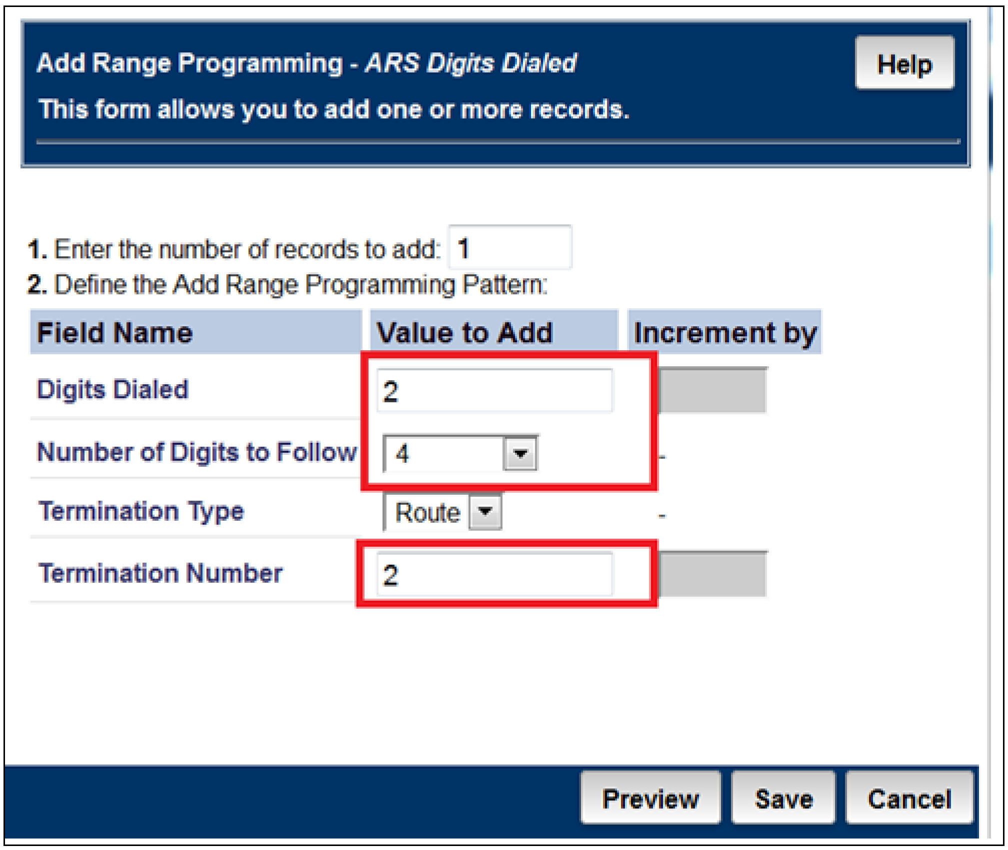
- In the **Routing Medium** drop down box select "SIP Trunk".
- In the **SIP Peer Profile** select the peer created in **Section 6.5**.
- Enter the COR Group Number created in **Section 6.7**.
- Enter the Digit Modification Number created in **Section 6.8**.



6.10 Program the ARS Digits Dialed Form

Navigate to **Call Routing → Automatic Route Selection (ARS) → ARS Digits Dialed**
In this example the Avaya extensions are 5 digits in length and begin with a 2.

- Program the **Digits Dialed** field with the 1st digit of Avaya extensions.
- Program the **Number of Digits to Follow** field to be the number of digits in the Avaya extension, minus 1 digit (the "2" programmed in Digits Dialed).
- Select **Route** for **Termination Type**.
- Program **Termination Number** to match the route created in **Section 6.9**.



6.11 Edit the Shared System Options Form.

Navigate to System Properties→System Feature Settings→Shared System Options
Verify that **DPNSS/QSIG Diversion Enabled** is set to **No**.
Note: This option must match on all cluster elements.

7 Verification Steps

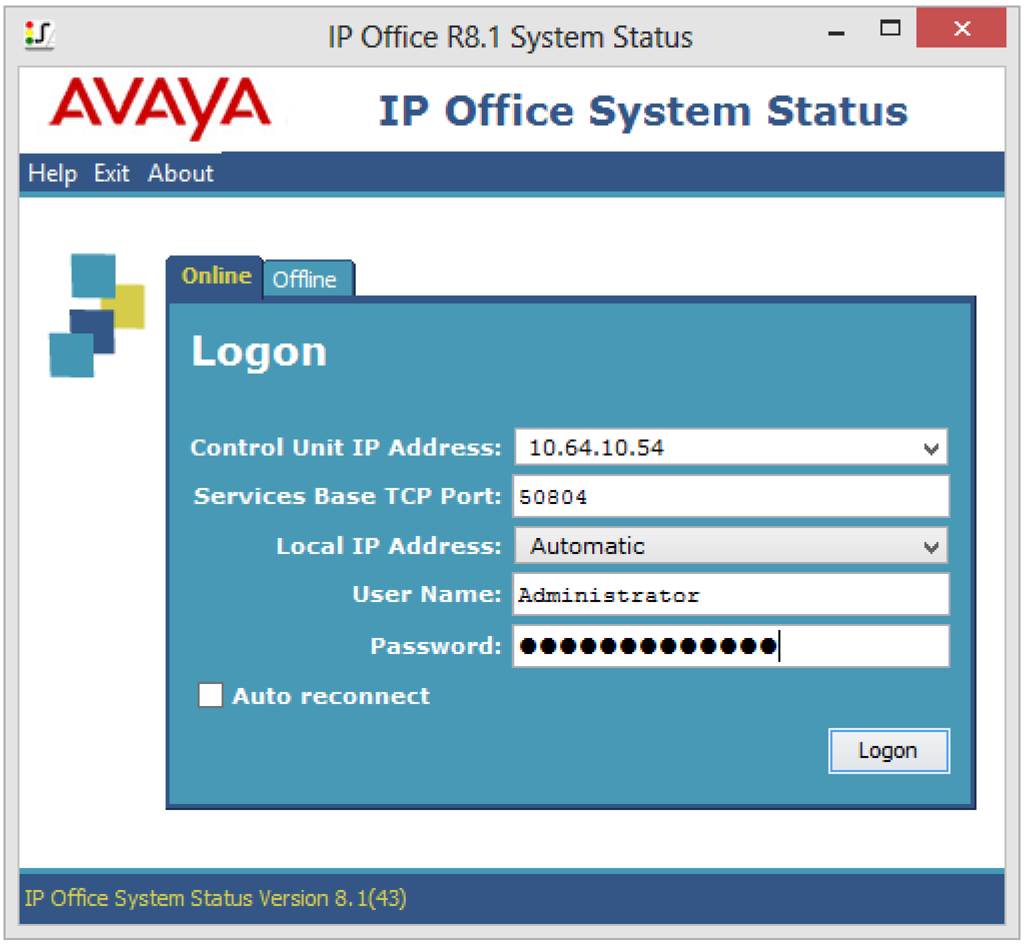
This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting tips that can be used for troubleshooting.

7.1 Avaya IP Office

The following steps may be used to verify the configuration:

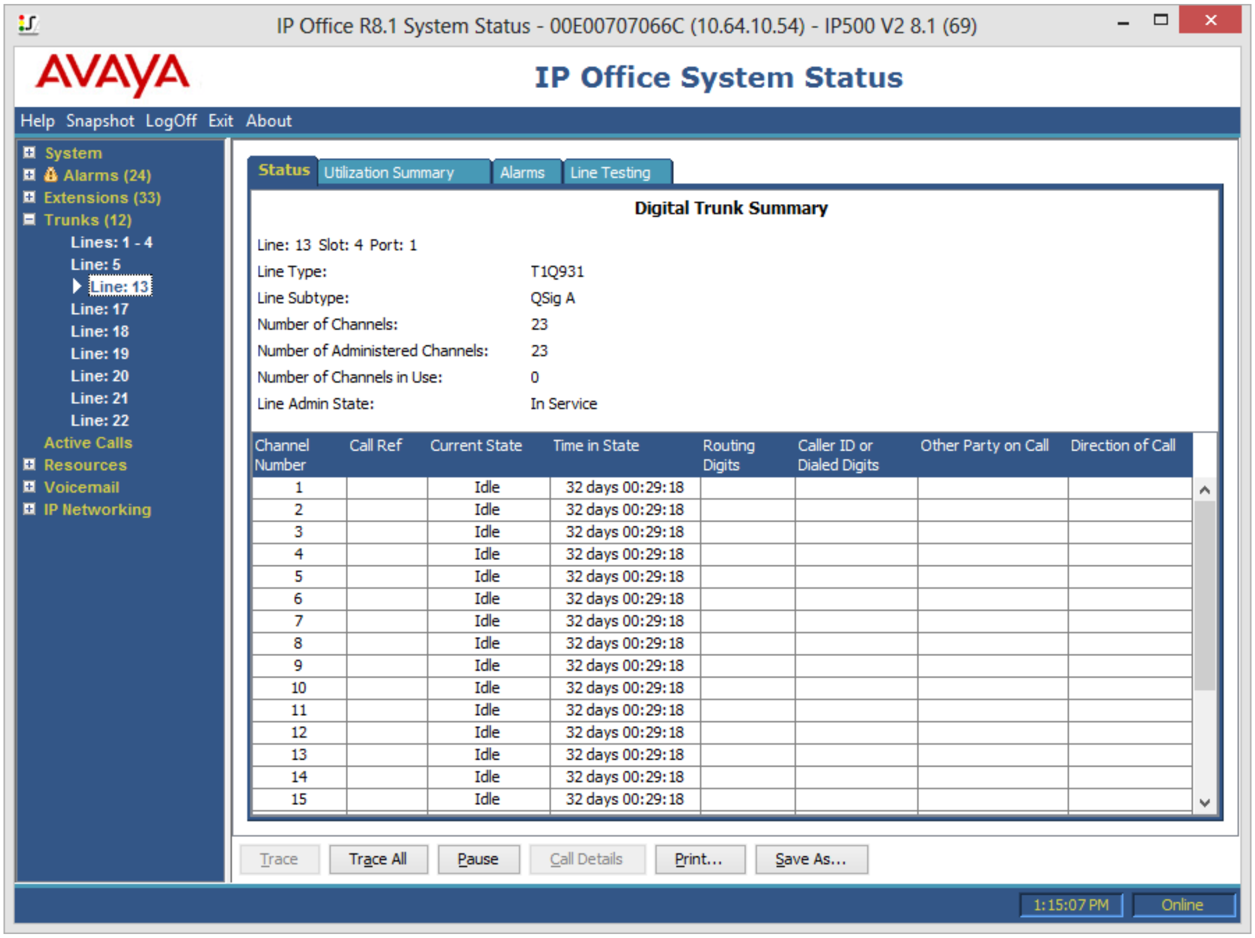
Using IP Office Manager, navigate to **File→Advanced→System Status** (Not shown). The following screen will be displayed.

Enter the appropriate credentials and click **Logon**.



Navigate to **Trunks → Line: *n*** in the left pane, where ***n*** is the line number of PRI line configured in this document. Select and verify the status of each trunk used in the configuration.

Current State of all **Channels** should be **idle.**



7.2 Wesley Clover Solutions

Navigate to **Maintenance and Diagnostic→Maintenance Commands**

The following maintenance commands may be useful for testing and validation. Please refer to the Wesley Clover Solutions IP PBX help files for additional commands and detailed descriptions.

- SIP LINK STATE ALL
  This command will show the UP/DOWN status of your SIP links.

- SIP ALL TRACE <ON/OFF>
  This command is used to start and stop SIP tracing directly to the following files: /db/SipTrace.rtf and /db/SipTrace_backup.rtf.
  The /db/SipTrace.rtf file may grow to a maximum size of 10 Mbytes before overwriting the backup file.

- DGT TRACE <number>
  This command is useful to validate outbound ARS routing.

- LOGS READ SMDR NEWEST <number>
  This command may be used to check call records for inbound or outbound calls.
  <number> is the number of records to read.

8. Conclusion

Wesley Clover Solutions Trading Platform passed compliance testing with one observation mentioned in **Section 2.2**. These Application Notes describe the procedures required to configure Wesley Clover Solutions Trading Platform to interoperate with Avaya IP Office to support the network shown in **Figure 1**.

9. Additional References

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

[1] Avaya IP Office 8.1 Installation,15-601042 Issue 26i – (23 August 2012)
[2] *Avaya IP Office R8.1 Manager, 10.115-601011 Issue 29o – (03 August 2012)*

Product information for Wesley Clover Solutions Trading Platform can be obtained from www.wesleycloversolutions.com

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.