



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.

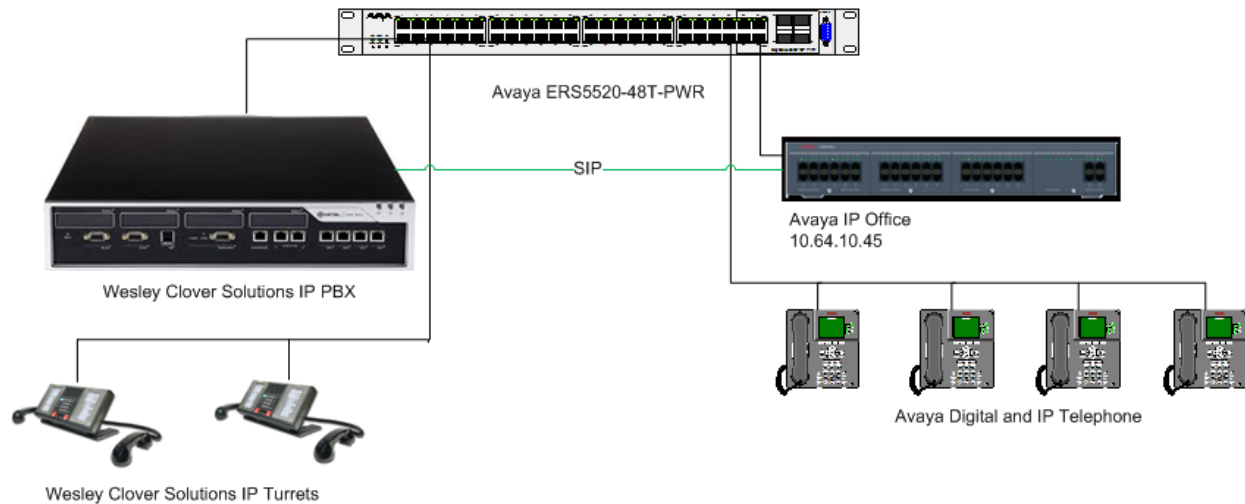


Figure 1: Reference Configuration for Wesley Clover Solutions IP PBX

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

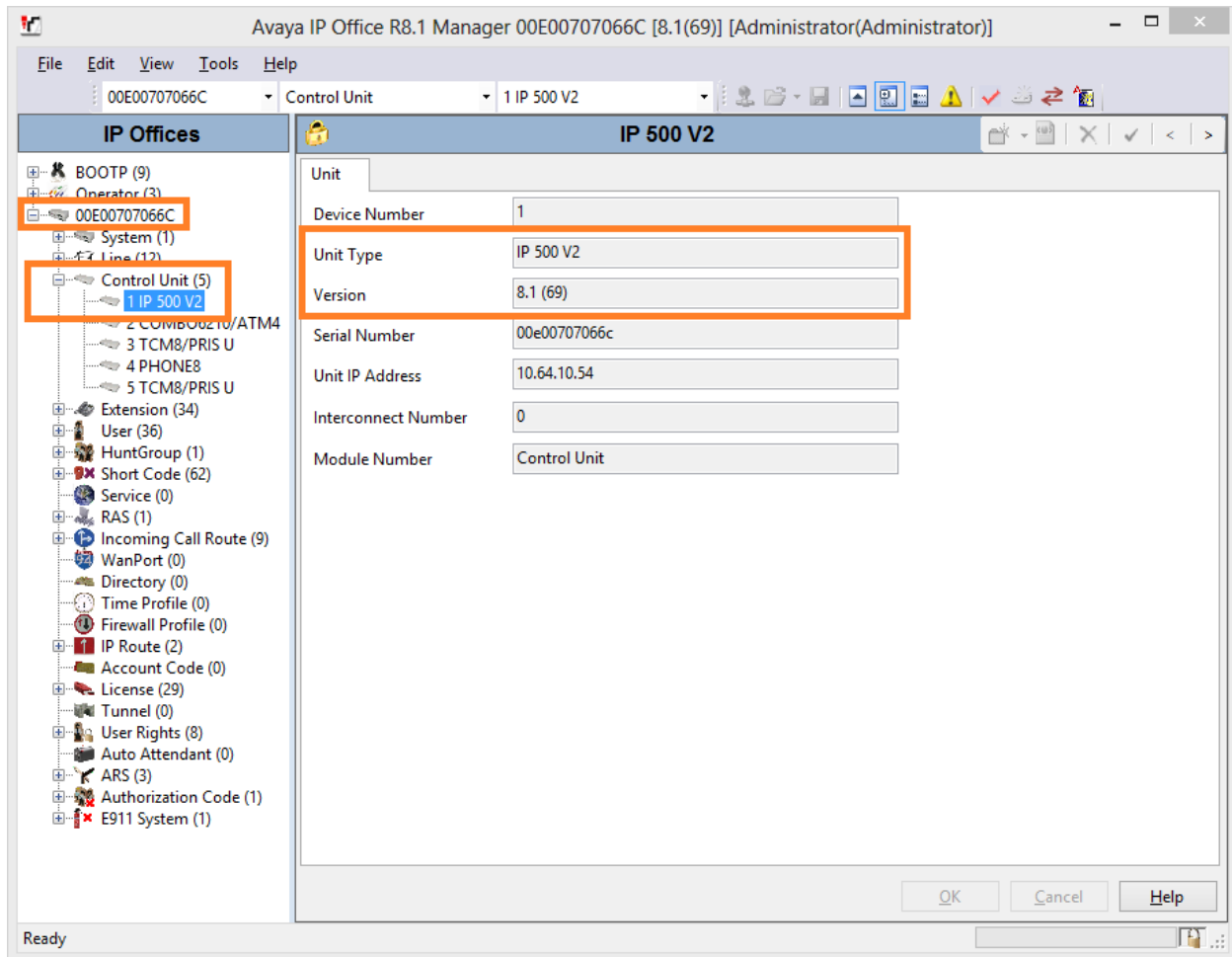
5.1 Connect to Avaya IP Office using Avaya IP Office Manager

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

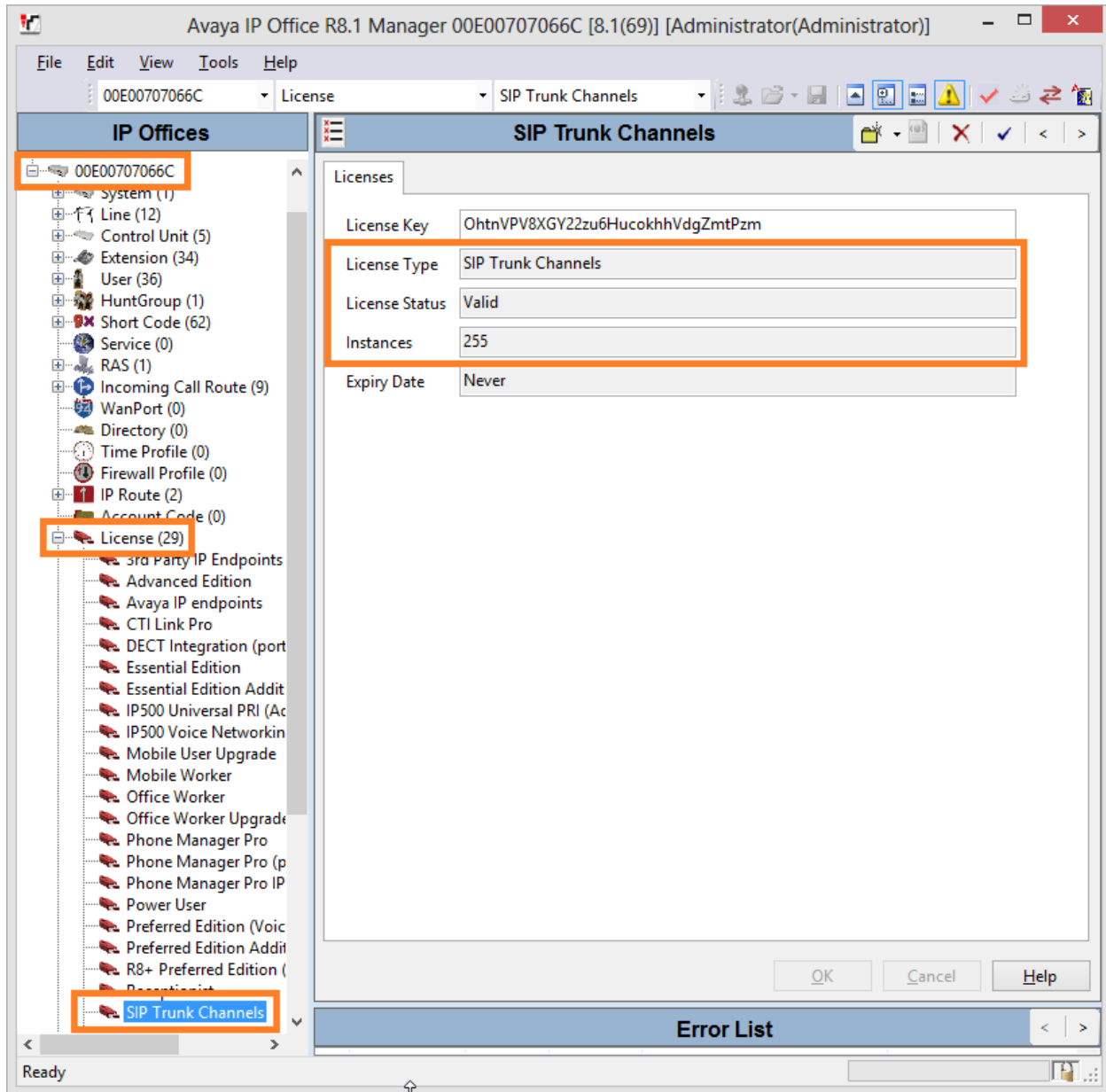
From the configuration tree in the navigation pane on the left, navigate to **IP 500 V2** as shown in the screen shot below.

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

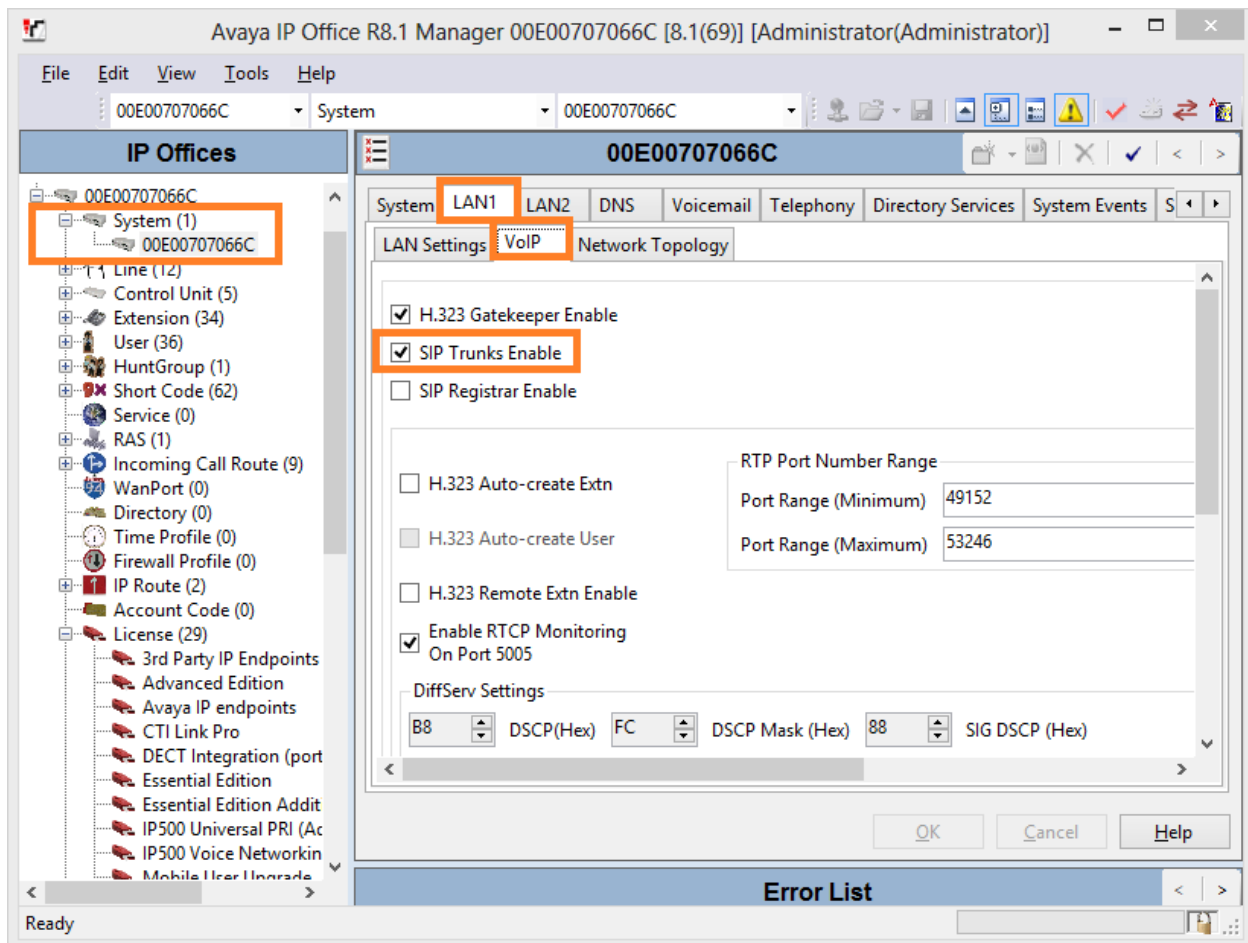
From the left pane, expand **System** then highlight **LAN1** in the right pane. From this page the **LAN Settings** and **VoIP** tabs will need to be configured.

5.4.1 LAN Settings

It is assumed that LAN Settings have already been configured.

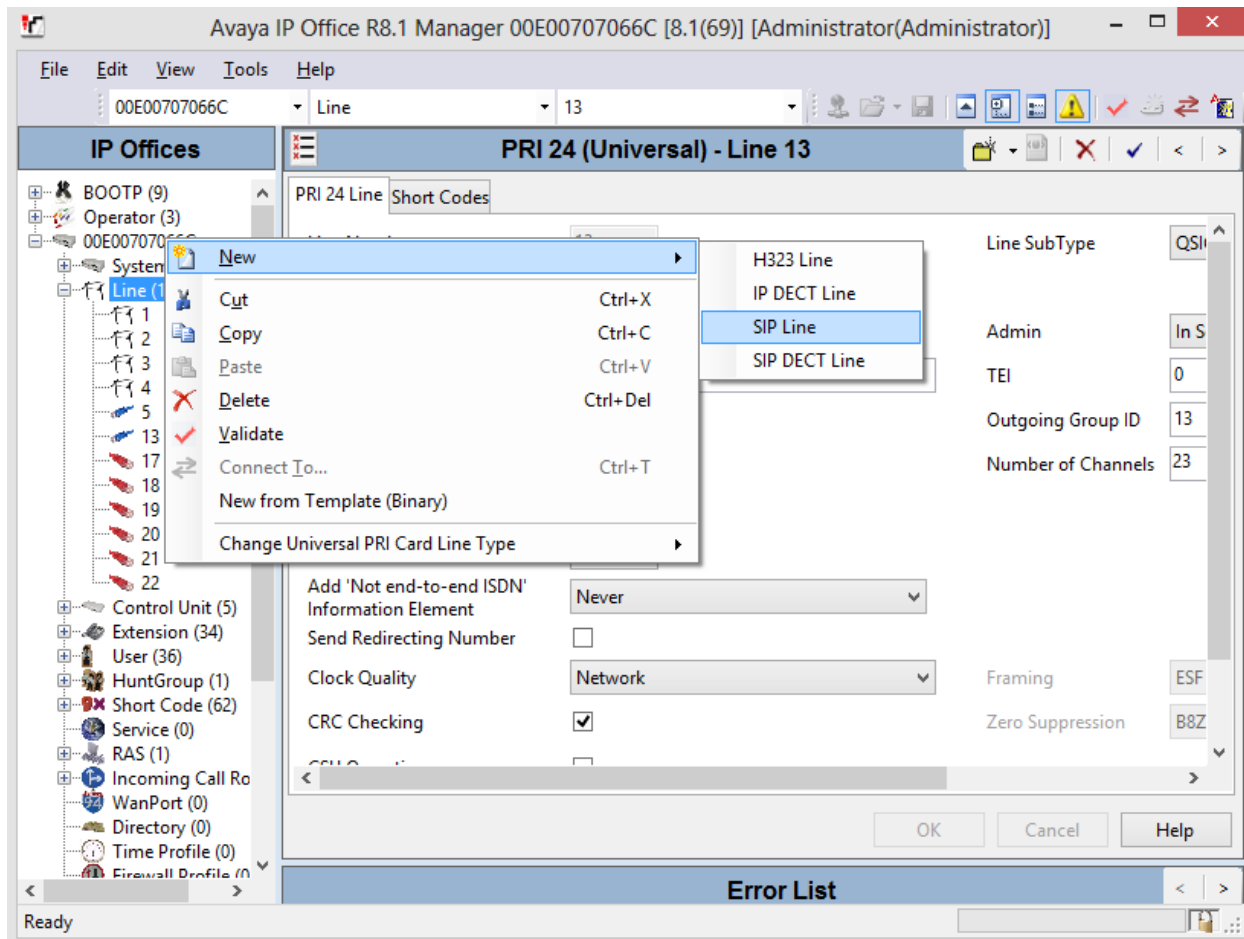
5.4.2 VoIP

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

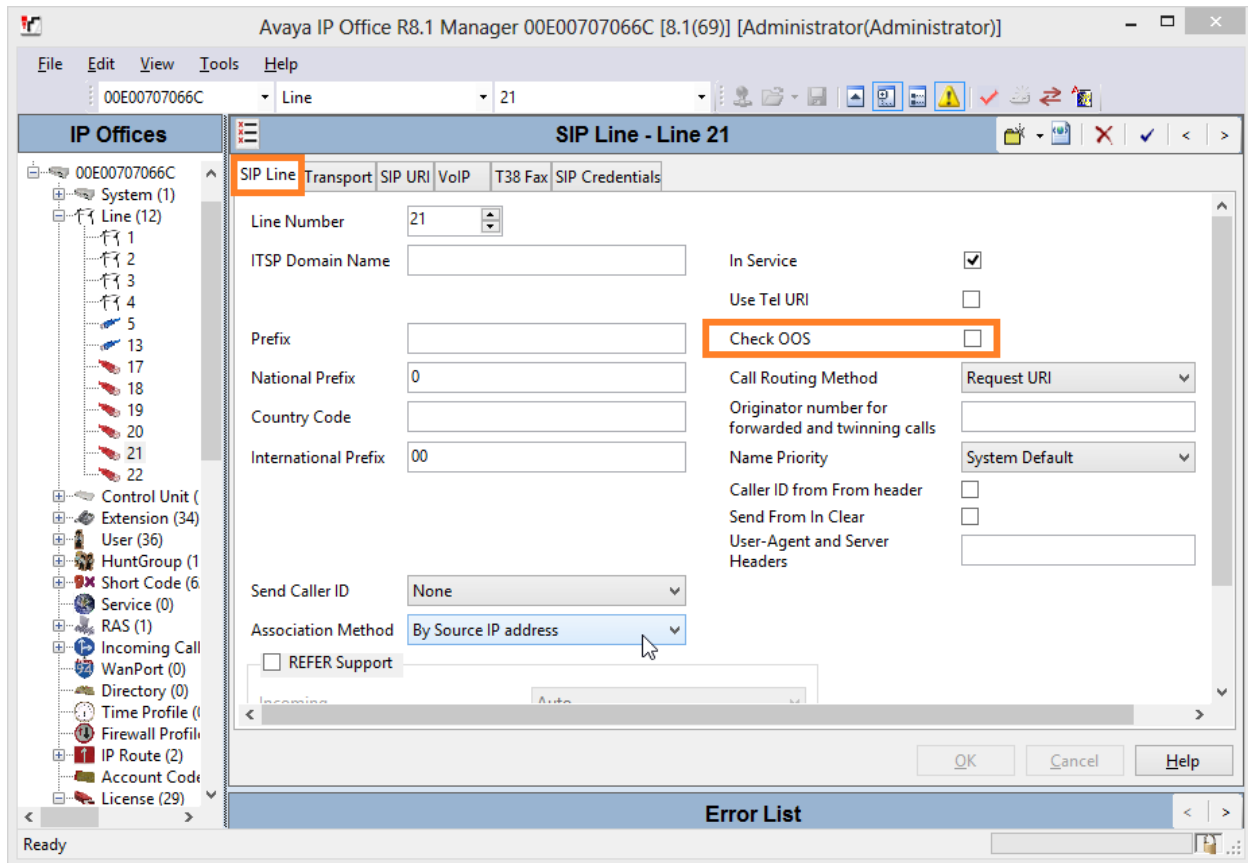


5.5 Configure Avaya IP Office SIP Line

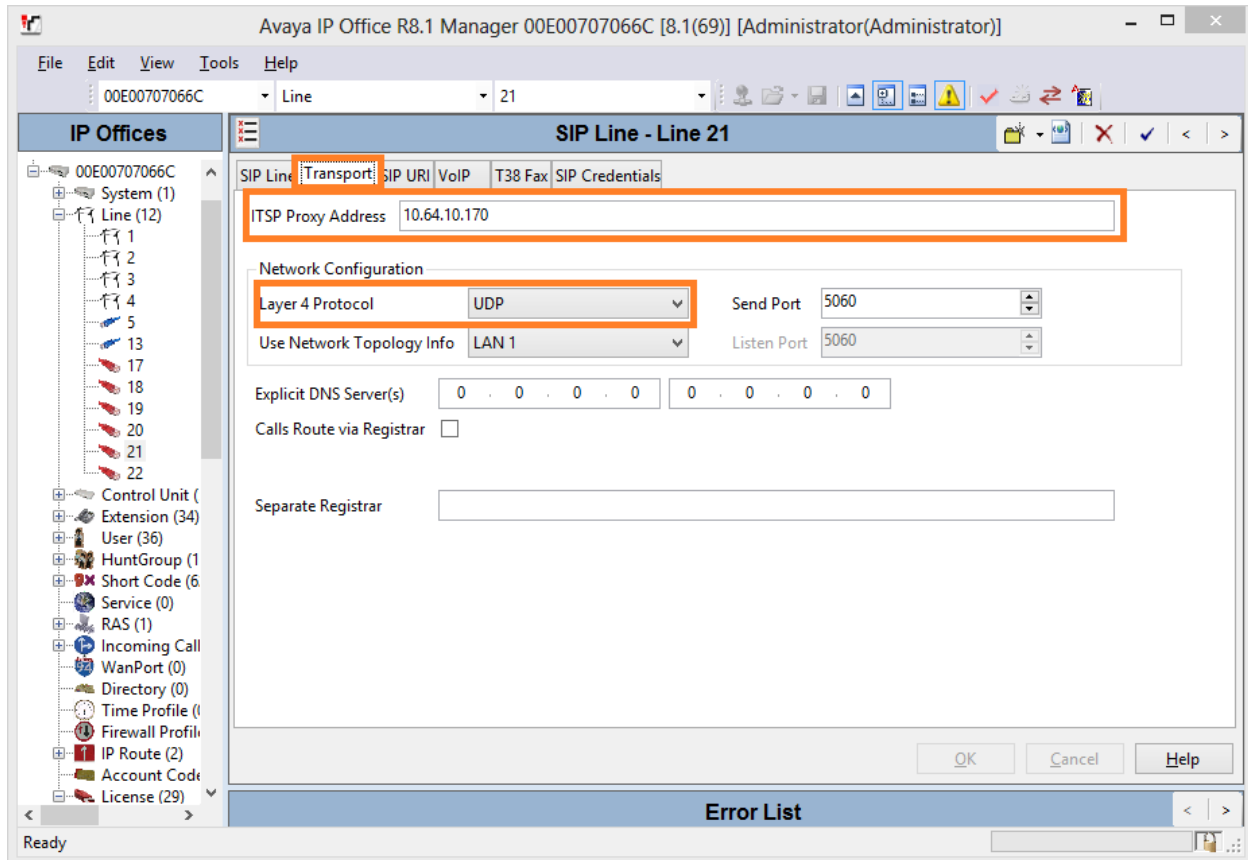
On the left pane, right click on **Line**, select **New** → **SIP Line**.



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

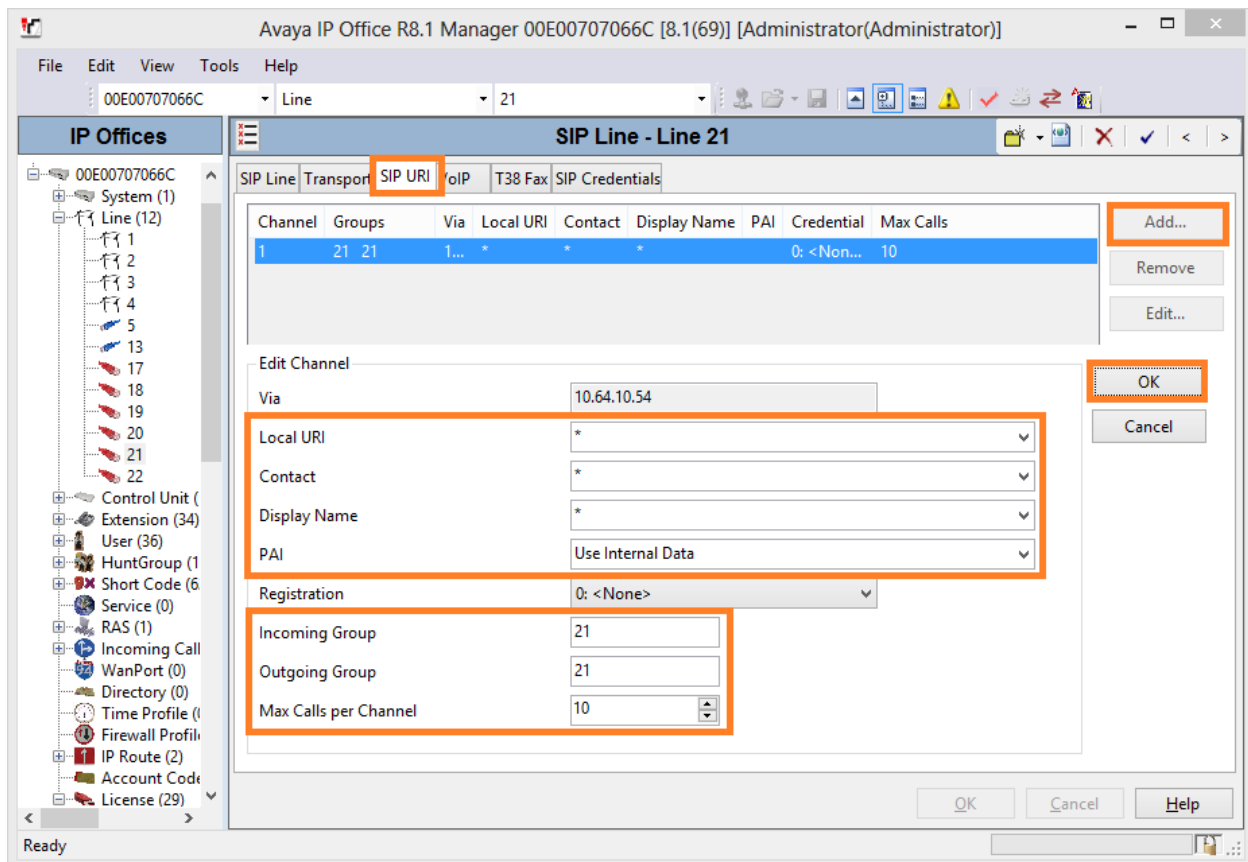


Under the **Transport** tab, in **ITSP Proxy Address**, type in the IP address of Wesley Clover Solutions IP PBX. Set **Layer 4 Protocol** to **UDP**.



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

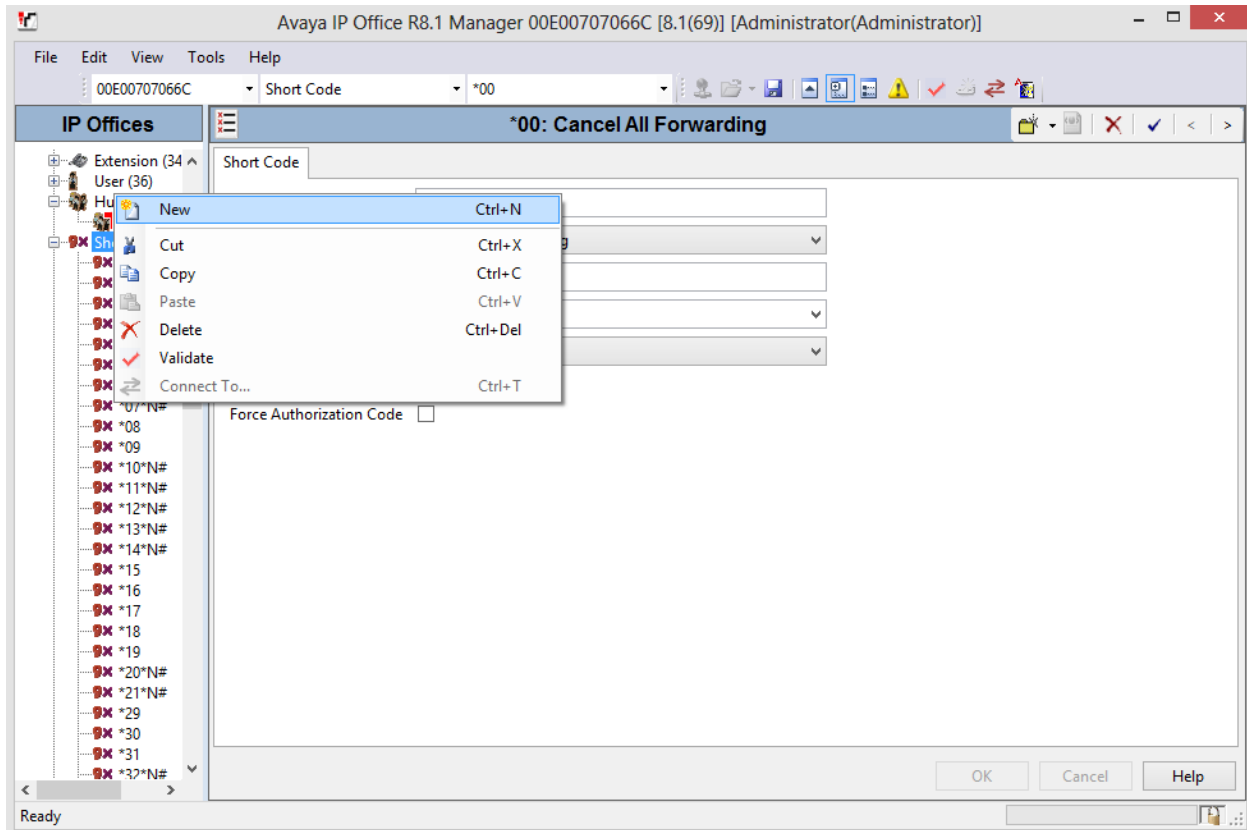
- Set **Local URI**, **Contact** and **Display Name** to *.
- In **Incoming Group** and **Outgoing Group**, type in the SIP Line number that was configured in **Section 5.5**.
- Set the number of **Max Calls per Channel**. During compliance testing, 10 **Max Calls per Channel** was used.



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:

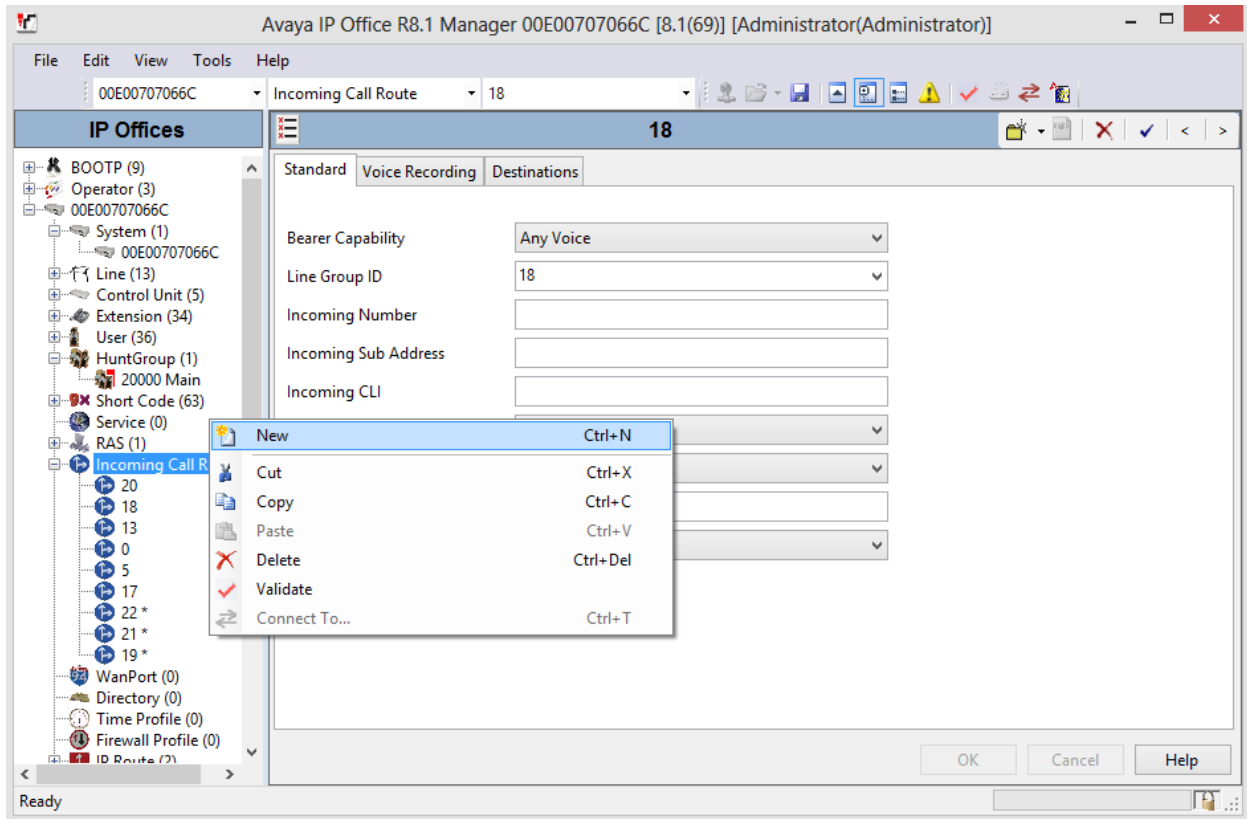
- Type in **4N;** in **Code** field.
- Set **Feature** to **Dial**.
- Type in **4N"@ip-address"** for **Telephone Number**, where ip-address is the IP Address of Wesley Clover Solutions IP PBX.
- Type in the number of PRI Line that was configured **Section 5.6**, in **Line Group ID** field.

The screenshot shows the Avaya IP Office R8.1 Manager interface. The title bar reads "Avaya IP Office R8.1 Manager 00E00707066C [8.1(69)] [Administrator/Administrator]". The main window has a menu bar (File, Edit, View, Tools, Help) and a toolbar. On the left, there is a tree view labeled "IP Offices" with a list of extensions: *37*N#, *38*N#, *39, *40, *41, *42, *43, *44, *45*N#, *46, *47, *48, *49, *50, *51, *52, *53*N#, *57*N#, *70*N#, *71*N#, *9000*, *91N;, *92N;, *DSSN, *SDN, *SKN, 250N;, 4N;, 4XXX, and 911. The main area is titled "4N;: Dial*" and contains a "Short Code" configuration form. The form fields are: Code (4N;), Feature (Dial), Telephone Number (4N"@10.64.10.170"), Line Group ID (21), and Locale (United States (US English)). There are also checkboxes for "Force Account Code" and "Force Authorization Code", both of which are unchecked. At the bottom right of the form are buttons for "OK", "Cancel", and "Help". The status bar at the bottom left shows "Ready".

Short Code	
Code	4N;
Feature	Dial
Telephone Number	4N"@10.64.10.170"
Line Group ID	21
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

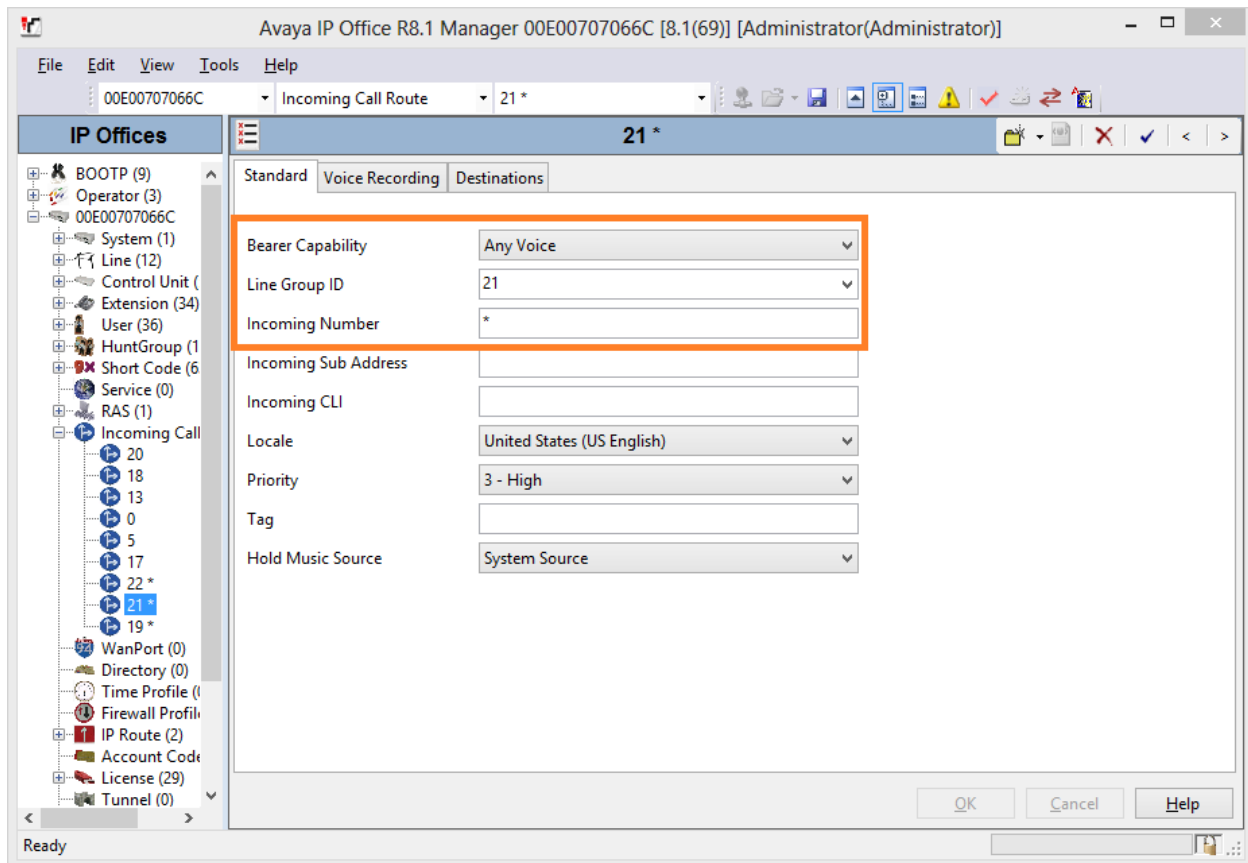
5.7 Configure Incoming Call Routes

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



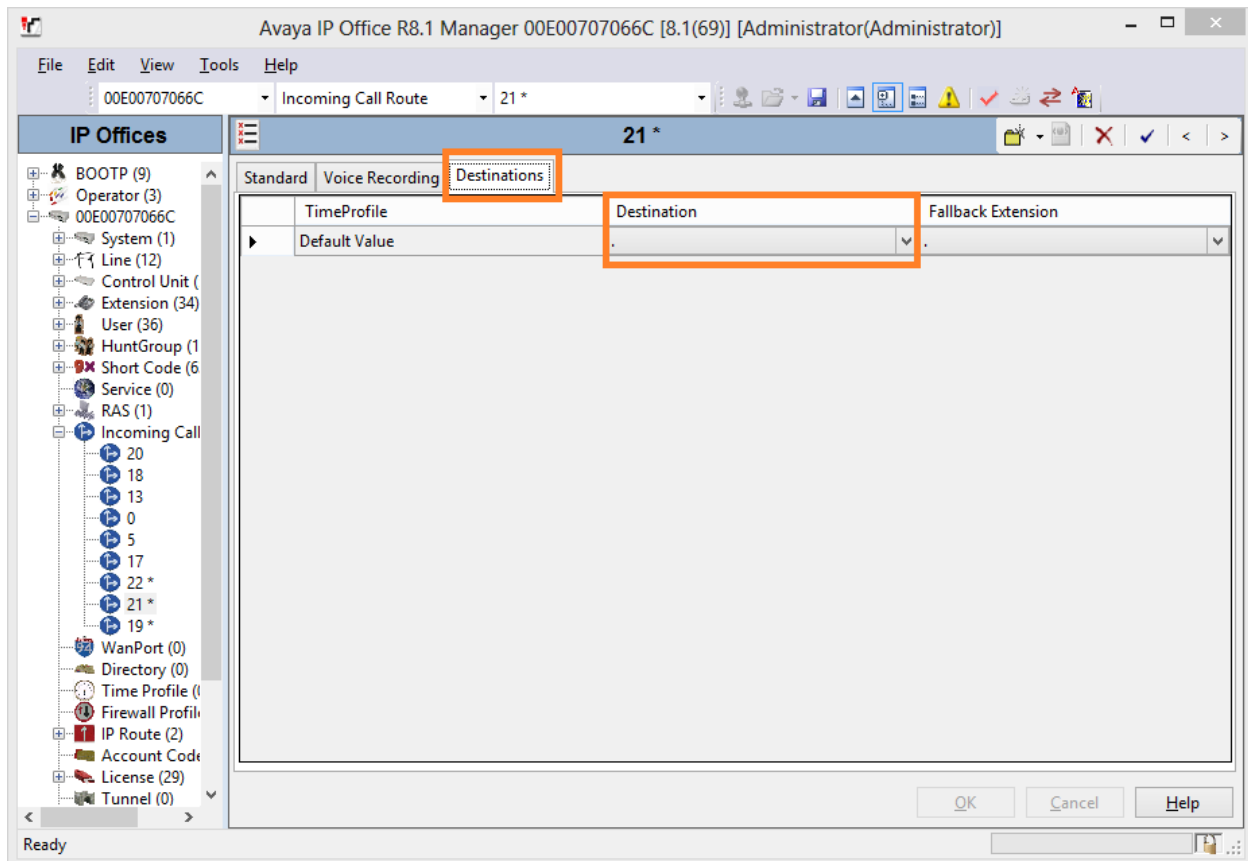
Under the **Standard** tab, configure the following:

- 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

Trunk	
ANI/DNIS/ISDN Number Delivery Trunk	<input checked="" type="radio"/> No <input type="radio"/> Yes
DASS II OLI/TLI Provided	<input checked="" type="radio"/> No <input type="radio"/> Yes
Public Network Access via DPNSS	<input type="radio"/> No <input checked="" type="radio"/> Yes
Public Network To Public Network Connection Allowed	<input type="radio"/> No <input checked="" type="radio"/> Yes
Public Trunk	<input type="radio"/> No <input checked="" type="radio"/> Yes
R2 Call Progress Tone	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Simulated CCM after ISDN Progress	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk Calling Party Identification	<input type="radio"/> No <input checked="" type="radio"/> Yes
Trunk Flash Allowed	<input checked="" type="radio"/> No <input type="radio"/> Yes
Two B-Channel Transfer Allowed	<input type="radio"/> No <input checked="" type="radio"/> Yes

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	<input checked="" type="radio"/> No <input type="radio"/> 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.

Name	Avaya
Type	Other
FQDN or IP Address	10.64.10.62
Local	False
Version	
Zone	1
ARID	
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	default
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal
<div>Save Cancel</div>	

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.

Network Elements	
Name	AvayaPrxy
Type	Outbound Proxy
FQDN or IP Address	10.64.10.46
Local	False
Version	
Zone	1
ARID	
Outbound Proxy Specific	
Outbound Proxy Transport Type	default
Outbound Proxy Port	5060
<div>Save Cancel</div>	

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**.

Basic		Call Routing	Calling Line ID	SDP Options	Signaling a
SIP Peer Profile Label	Avaya				
Network Element	Avaya				
Local Account Information					
Registration User Name					
Address Type	IP Address: 10.64.10.170				
Administration Options					
Interconnect Restriction	1				
Maximum Simultaneous Calls	5				
Minimum Reserved Call Licenses	0				
Administration Options					
Outbound Proxy Server	AvayaPrxy				
SMDR Tag	0				
Trunk Service	6				
Zone	1				
User Name					
Password	*****				
Confirm Password	*****				
Authentication Option for Incoming Calls	No Authentication				
Subscription User Name					
Subscription Password	*****				
Subscription Confirm Password	*****				

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.

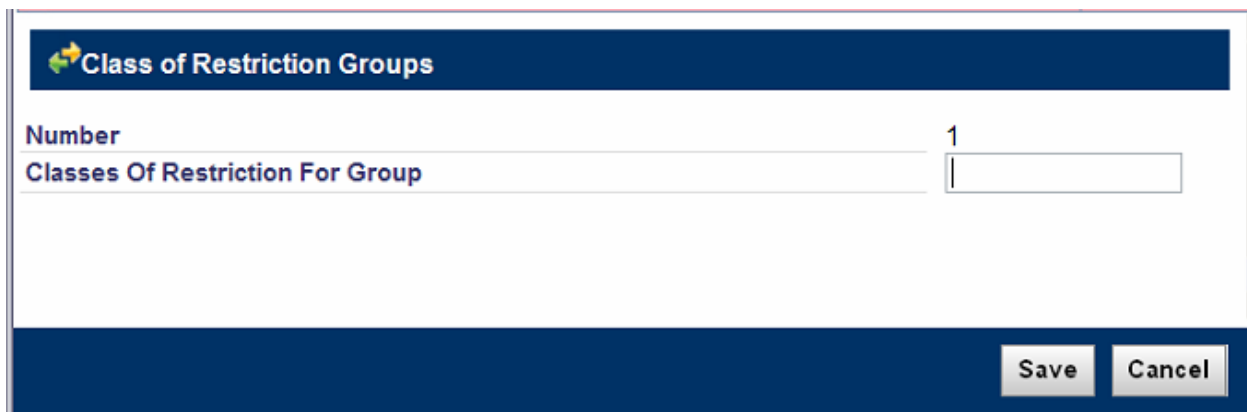


The screenshot shows a web form titled "SIP Peer Profile Assignment by Incoming DID". It contains three input fields: "Incoming DID Range" with the value "4000-5002", "SIP Peer Profile Label" with a dropdown menu showing "Avaya", and "Comment" with the text "Avaya SIP". The form is enclosed in a red rectangular border.

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.



The screenshot shows a web form titled "Class of Restriction Groups". It contains two input fields: "Number" with the value "1" and "Classes Of Restriction For Group" which is empty. At the bottom right, there are "Save" and "Cancel" buttons.

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**.

The screenshot shows the 'ARS Digit Modification Plans' form. It has a dark blue header with a small icon and the title. Below the header, there are four input fields: 'Digit Modification Number' with the value '1', 'Number of Digits to Absorb' with the value '0', 'Digits to be Inserted' (empty), and 'Final Tone Plan/Information Marker' (empty). At the bottom right, there are 'Save' and 'Cancel' buttons.

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**.

The screenshot shows the 'ARS Routes' form. It has a dark blue header with the title. Below the header, there are several input fields: 'Route Number' with the value '1', 'Routing Medium' with a dropdown menu showing 'SIP Trunk', 'Trunk Group Number' (empty), 'SIP Peer Profile' with a dropdown menu showing 'Avaya', 'PBX Number / Cluster Element ID' (empty), 'COR Group Number' with the value '1', 'Digit Modification Number' with the value '1', 'Digits Before Outpulsing' with a dropdown menu showing a small 'v' icon, 'Route Type' with a dropdown menu showing a small 'v' icon, and 'Compression' with a dropdown menu showing 'Off'. At the bottom right, there are 'Save' and 'Cancel' buttons.

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**.

Field Name	Value to Add	Increment by
Digits Dialed	2	
Number of Digits to Follow	4	
Termination Type	Route	
Termination Number	2	

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.

DPNSS/QSIG Diversion Enabled	No
Maintain Original Forward or Reroute Reason	No

7 Verification Steps

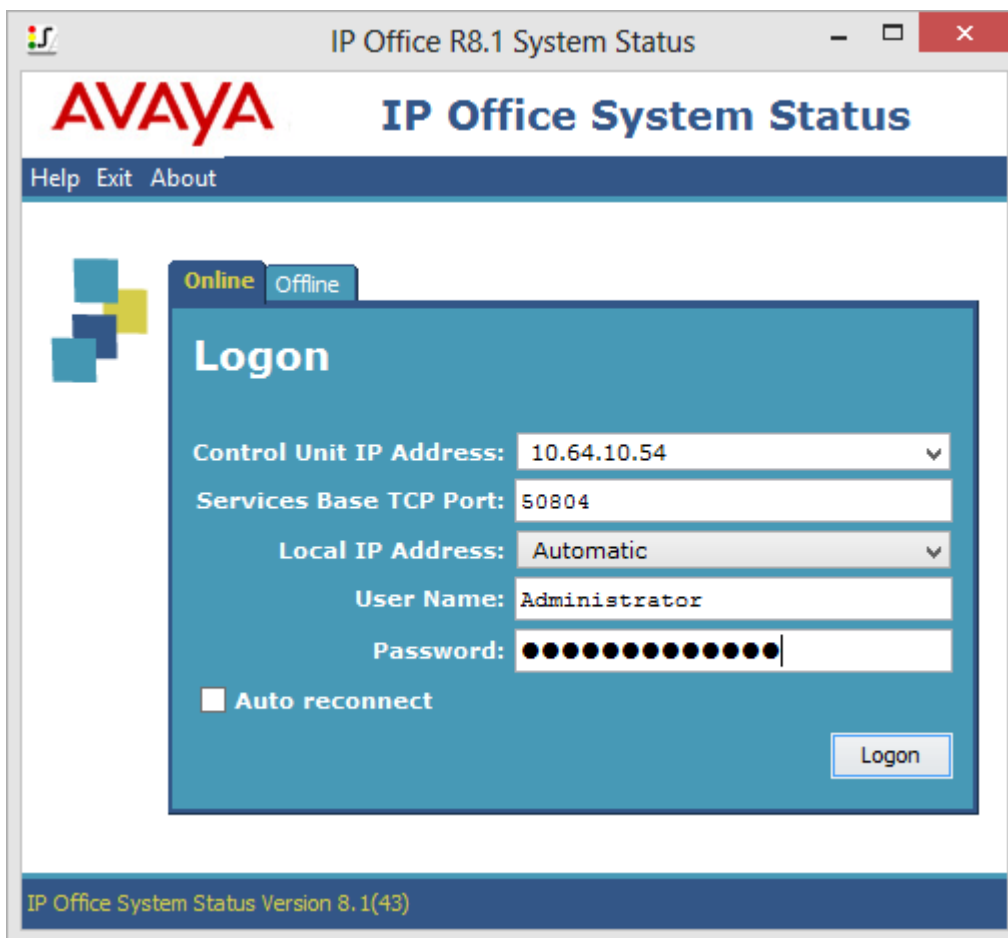
This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting tips that can be used for troubleshooting.

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**.



The screenshot shows the 'IP Office R8.1 System Status' window. The title bar includes the Avaya logo and the text 'IP Office R8.1 System Status'. The main window has a blue header with the 'AVAYA' logo and the title 'IP Office System Status'. Below the header is a menu bar with 'Help', 'Exit', and 'About'. The main content area features a 'Logon' dialog box. The dialog box has two tabs: 'Online' (selected) and 'Offline'. It contains the following fields: 'Control Unit IP Address' (10.64.10.54), 'Services Base TCP Port' (50804), 'Local IP Address' (Automatic), 'User Name' (Administrator), and 'Password' (masked with dots). There is an 'Auto reconnect' checkbox and a 'Logon' button. The status bar at the bottom of the window reads 'IP Office System Status Version 8.1(43)'.

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**.

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

System
Alarms (24)
Extensions (33)
Trunks (12)
 Lines: 1 - 4
 Line: 5
Line: 13
 Line: 17
 Line: 18
 Line: 19
 Line: 20
 Line: 21
 Line: 22
Active Calls
Resources
Voicemail
IP Networking

Status Utilization Summary Alarms Line Testing

Digital Trunk Summary

Line: 13 Slot: 4 Port: 1
 Line Type: T1Q931
 Line Subtype: QSig A
 Number of Channels: 23
 Number of Administered Channels: 23
 Number of Channels in Use: 0
 Line Admin State: In Service

Channel Number	Call Ref	Current State	Time in State	Routing Digits	Caller ID or Dialed Digits	Other Party on Call	Direction of Call
1		Idle	32 days 00:29:18				
2		Idle	32 days 00:29:18				
3		Idle	32 days 00:29:18				
4		Idle	32 days 00:29:18				
5		Idle	32 days 00:29:18				
6		Idle	32 days 00:29:18				
7		Idle	32 days 00:29:18				
8		Idle	32 days 00:29:18				
9		Idle	32 days 00:29:18				
10		Idle	32 days 00:29:18				
11		Idle	32 days 00:29:18				
12		Idle	32 days 00:29:18				
13		Idle	32 days 00:29:18				
14		Idle	32 days 00:29:18				
15		Idle	32 days 00:29:18				

Trace Trace All Pause Call Details Print... Save As...

1:15:07 PM Online

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

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