



Application Notes for Wesley Clover Solutions Trading Platform with Avaya Aura® Session Manager and Avaya Aura® Communication Manager – Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration consisting of Wesley Clover Solutions IP PBX and Wesley Clover Solutions IP Turret with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

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 Aura® Session Manager via a SIP trunk using 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 Aura® Session Manager 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
- Call setup using G711MU, G711A and G729A
- DTMF support using RFC2833 and in-band
- Call Holds, 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 Transfers, Conferences and Call Forwards, Wesley Clover Solutions IP PBX holds onto SIP resource for each call leg

2.3. Support

Support for Wesley Clover Solutions can be found via the following means:

Web: www.wesleycloversolutions.com

E-mail: service@wesleycloversolutions.com

3. Reference Configuration

The following figure displays the configuration was used during the compliance test.

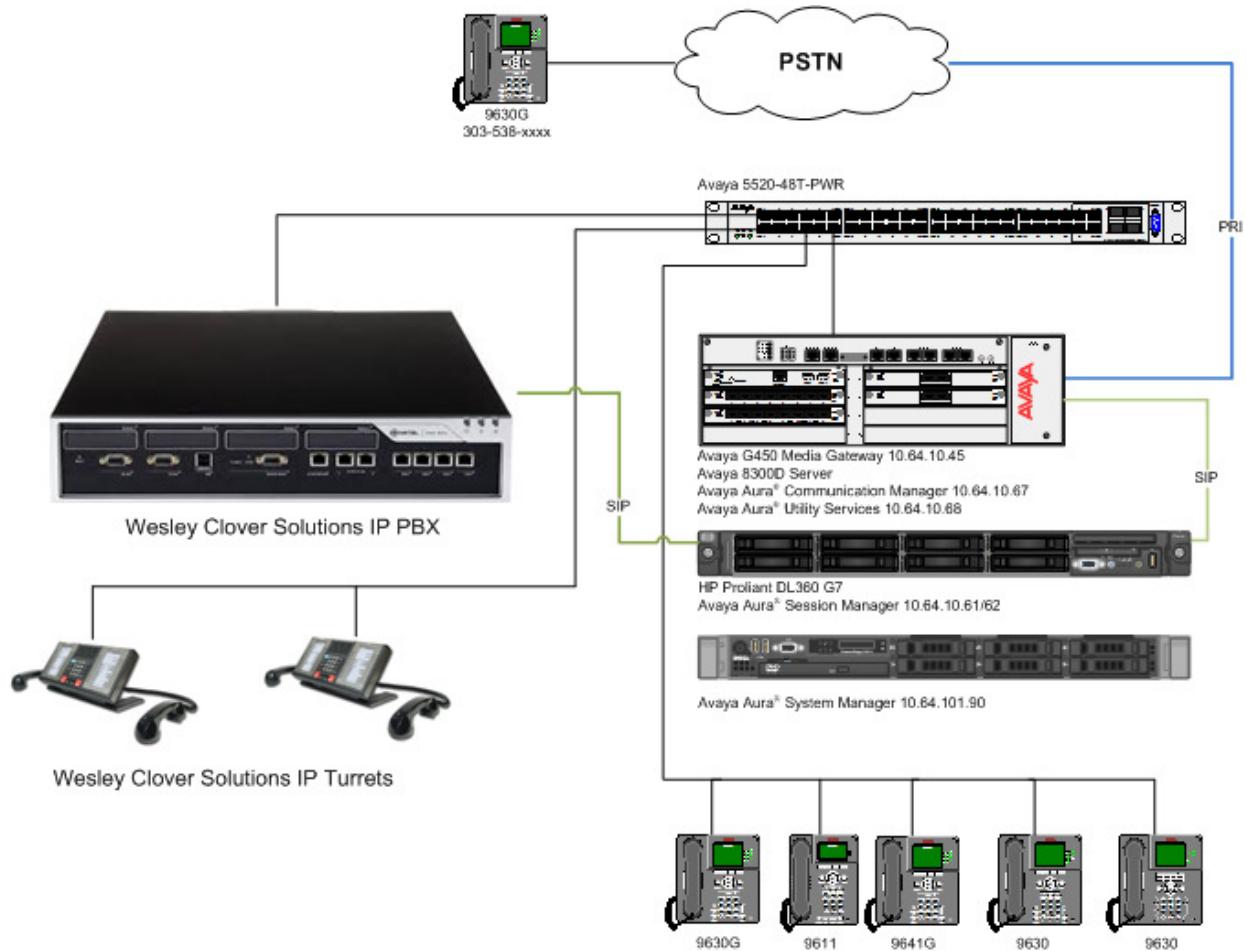


Figure 1: Reference Configuration

4. Equipment and Software Validated

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

Equipment	Version
Avaya Aura [®] System Manager	6.3.3 (SP3)
Avaya Aura [®] Session Manager	6.3.3
Avaya Aura [®] Communication Manager	6.3.1 (SP1)
Avaya G450 Media Gateway	31.20.1
Avaya 96x1 Series H.323 Phones	6.2.4
Avaya 96x0 Series H.323 Phones	3.10
Avaya Aura [®] Utility Services	6.3.1 (SP1)
Wesley Clover Solutions IP PBX	12.0.1.24
Wesley Clover Solutions IP Turrets	3.0.0.8

5. Configure Avaya Aura® Communication Manager

Communication Manager allows for routing calls to a SIP trunk to Wesley Clover Solutions IP PBX via Session Manager. The following information allows for a SIP connection between Communication Manager and Session Manager.

5.1. Verify Avaya Aura® Communication Manager License

Use the **display system-parameters customer-options** command to verify options.

On **Page 2**, verify that there is sufficient capacity for SIP trunks by comparing **Maximum Administered SIP Trunks** field with corresponding **USED** column field.

display system-parameters customer-options		Page	2	of	11
OPTIONAL FEATURES					
IP PORT CAPACITIES		USED			
	Maximum Administered H.323 Trunks:	4000	0		
	Maximum Concurrently Registered IP Stations:	2400	1		
	Maximum Administered Remote Office Trunks:	4000	0		
	Maximum Concurrently Registered Remote Office Stations:	2400	0		
	Maximum Concurrently Registered IP eCons:	68	0		
	Max Concur Registered Unauthenticated H.323 Stations:	100	0		
	Maximum Video Capable Stations:	2400	0		
	Maximum Video Capable IP Softphones:	2400	0		
	Maximum Administered SIP Trunks:	4000	45		
	Maximum Administered Ad-hoc Video Conferencing Ports:	4000	0		
	Maximum Number of DS1 Boards with Echo Cancellation:	80	0		
	Maximum TN2501 VAL Boards:	10	0		
	Maximum Media Gateway VAL Sources:	50	0		
	Maximum TN2602 Boards with 80 VoIP Channels:	128	0		
	Maximum TN2602 Boards with 320 VoIP Channels:	128	0		
	Maximum Number of Expanded Meet-me Conference Ports:	300	0		

5.2. Configure IP Network Region

Use the **change ip-network-region *n*** command to configure a network region, where *n* is an existing network region. Configure this network region as follows:

- Set **Location** to **1**
- Set **Codec Set** to **1**
- Set **Intra-region IP-IP Direct Audio** to **yes**
- Set **Inter-region IP-IP Direct Audio** to **yes**
- Enter and **Authoritative Domain**, e.g., **avaya.com**

```
change ip-network-region 1                                     Page 1 of 20
                                                              IP NETWORK REGION
    Region: 1
    Location: 1          Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                      Intra-region IP-IP Direct Audio: yes
    Codec Set: 1          Inter-region IP-IP Direct Audio: yes
    UDP Port Min: 2048          IP Audio Hairpinning? n
    UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
    Call Control PHB Value: 46
    Audio PHB Value: 46
    Video PHB Value: 26
802.1P/Q PARAMETERS
    Call Control 802.1p Priority: 6
    Audio 802.1p Priority: 6
    Video 802.1p Priority: 5
H.323 IP DESKPHONES          AUDIO RESOURCE RESERVATION PARAMETERS
    H.323 Link Bounce Recovery? y          RSVP Enabled? n
    Idle Traffic Interval (sec): 20
    Keep-Alive Interval (sec): 5
```

5.3. Administer IP Codec Set

Use the **change ip-codec-set *n*** command to configure IP codec set, where *n* is an existing codec set number. Configure this codec set as follows, on **Page 1**:

- Set Audio Codec 1 to **G.711MU**

change ip-codec-set 1				Page 1 of 2
IP Codec Set				
Codec Set: 1				
Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)	
1: G.711MU	n	2	20	
2: G.711A	n	2	20	
3: G.729AB	n	2	20	
4:				
5:				
6:				
7:				

5.4. Administer IP Node Names

Use the **change node-names ip** command to add an entry for Session Manager. For compliance testing, **sm** and **10.64.10.62** entry was added.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
default	0.0.0.0	
msgsrvr	192.168.62.28	
procr	192.168.62.28	
procr6	::	
sm	10.64.10.62	

5.5. Administer SIP Signaling Group

Use the **add signaling-group *n*** command to add a new signaling group, where *n* is an available signaling group number. Configure this signaling group as follows:

- Set **Group Type** to **sip**
- Set **Near-end Node Name** to **procr**
- Set **Far-end Node Name** to the configured Session Manager in **Section 5.3**, e.g., **sm**
- Set **Far-end Network region** to the configured region in **Section 5.1**, e.g., **1**
- Enter a **Far-end Domain**, e.g., **avaya.com**

add signaling-group 10		Page 1 of 2
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: Others	
Near-end Node Name: procr	Far-end Node Name: sm	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain: avaya.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

5.6. Administer SIP Trunk Group

Use the **add trunk-group *n*** command to add a trunk group, where *n* is an available trunk group number. Configure this trunk group as follows, on **Page 1**:

- Set **Group Type** to **sip**
- Enter a **Group Name**
- Enter a valid **TAC**, e.g., *010
- Set **Service Type** to **tie**
- Enter **Signaling Group** value to the signaling group configured in **Section 5.5**, e.g., 10
- Enter a desired number in **Number of Members** field

add trunk-group 10		Page 1 of 21	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: Session Manager	COR: 1	TN: 1	TAC: *010
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group: 10		
	Number of Members: 25		

On **Page 3**:

- Set **Number Format** to **private**

add trunk-group 10		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private		UI Treatment: service-provider	
Replace Restricted Numbers? n			
Replace Unavailable Numbers? n			

5.7. Administer Route Pattern

Use the **change route-pattern *n*** command to configure a route pattern, where *n* is an available route pattern. Configure this route pattern as follows:

- Type a name in **Pattern Name** field
- For line 1, set **Grp No** to the trunk group configured in **Section 5.6**, e.g., 10
- For line 1, set **FRL** to **0**

change route-pattern 10											Page	1	of	3
Pattern Number: 1											Pattern Name: Voice and Fax			
SCCAN? n											Secure SIP? n			
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/ IXC						
No			Mrk	Lmt	List	Del	Digits	QSIG						
								Intw						
1:	10	0						n user						
2:						n user								

5.8. Administer Private Numbering

Use the **change private-numbering 1** command to define the calling party number to send to Session Manager.

Configure private numbering as follows:

- Add entries for the trunk group configured in **Section 5.6**

change private-numbering 1										Page	1	of	2
NUMBERING - PRIVATE FORMAT													
Ext	Ext			Trk			Private			Total			
Len	Code			Grp(s)			Prefix			Len			
5	2			10						5	Total Administered: 1		
5	5									5	Maximum Entries: 540		

5.9. Administer AAR Analysis

Use the **change aar analysis *n*** command to configure routing for extensions starting with *n*. For compliance testing, extensions starting with 4 and 4 digits long were used to route calls to Wesley Clover IP PBX:

- Set **Dialed String** to starting digits of extensions that will be used, e.g., 4
- Set **Min** and **Max** to 4 for 4 digit extensions
- Set **Route Pattern** to pattern configured in **Section 5.6**, e.g., 10
- Set **Call Type** to **aar**

Note: An entry to dial plan will need to be added for extension range used in this step.

change aar analysis 4						Page 1 of 2	
AAR DIGIT ANALYSIS TABLE							
Location: all				Percent Full: 1			
	Dialed String	Total Min	Max	Route Pattern	Call Type	Node Num	ANI Req'd
4		4	4	10	aar		n
4		5	5	2	aar		n
45000		5	5	30	aar		n
5		4	4	2	lev0		n
5		5	5	32	aar		n
552		10	10	10	aar		n
588		5	5	10	aar		n

5.10. Administer Stations

Administration of Avaya Stations/Extensions in Communication Manager and Session Manager is not shown in this document. Please refer to document [1] and/or [2] in reference section of this document.

6. Configure Avaya Aura® Session Manager

Access the Session Manager Administration web interface by entering <https://<ip-address>/SMGR> URL in a web browser, where <ip-address> is the IP address of System Manager.



Avaya Aura® System Manager 6.3

[Home / Log On](#)

Log On

Recommended access to System Manager is via FQDN.

[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

User ID:

Password:

[Change Password](#)

Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 15.0, 16.0 or 17.0.

Log in using appropriate credentials.



Users	Elements	Services
Administrators Manage Administrative Users	Communication Manager Manage Communication Manager 5.2 and higher elements	Backup and Restore Backup and restore System Manager database
Directory Synchronization Synchronize users with the enterprise directory	Communication Server 1000 Manage Communication Server 1000 elements	Bulk Import and Export Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
Groups & Roles Manage groups, roles and assign roles to users	Conferencing Manage Conferencing Multimedia Server objects	Configurations Manage system wide configurations
User Management Manage users, shared user resources and provision users	IP Office Manage IP Office elements	Events Manage alarms, view and harvest logs
	Meeting Exchange Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements	Geographic Redundancy Manage Geographic Redundancy
	Messaging Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging	Inventory Manage, discover, and navigate to elements
	Presence Presence	Licenses View and configure licenses
	Routing Session Manager Routing Administration	Replication Track data replication nodes, repair replication nodes
	Session Manager Session Manager Administration, Status, Maintenance and Performance Management	Scheduler Schedule, track, cancel, update and delete jobs
		Security Manage Security Certificates
		Shutdown Shutdown System Manager Gracefully
		Software Management Upgrade and Patch Management for Communication Manager devices and IP Office

6.1. Add SIP Domain

Navigate to **Home → Elements → Routing → Domains**, click on **New** button (not shown) and configure as follows:

- In **Name** field type in a domain (authoritative domain used in **Section 5**) e.g., avaya.com
- Set **Type** to **sip**

Click **Commit** to save changes.

AVAYA Avaya Aura® System Manager 6.3

Last Logged on at July 25, 2013 9:29 AM
Help | About | Change Password | Log out | admin

Routing * Home

Home / Elements / Routing / Domains

Domain Management

Commit Cancel

1 Item | Refresh Filter: Enable

Name	Type	Notes
* avaya.com	sip	

Commit Cancel

6.2. Add Location

Navigate to **Home → Elements → Routing → Locations**, click on **New** button (not shown) and configure as follows:

Under **General**:

- Type in a descriptive **Name**

Under **Location Pattern** click on **New** (not shown):

- Type in an **IP Address Pattern**, e.g., 10.64.10.*

Click **Commit** to save changes. Screen shot shown on next page.

Location Details

[Commit](#) [Cancel](#)

General

* Name:

Notes:

Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units:

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): Kbit/Sec

* Minimum Multimedia Bandwidth: Kbit/Sec

* Default Audio Bandwidth:

Alarm Threshold

Overall Alarm Threshold: %

Multimedia Alarm Threshold: %

* Latency before Overall Alarm Trigger: Minutes

* Latency before Multimedia Alarm Trigger: Minutes

Location Pattern

[Add](#) [Remove](#)

2 Items Refresh		Filter: Enable
<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.64.10.*	<input type="text"/>
<input type="checkbox"/>	* 10.64.101.*	<input type="text"/>
Select : All, None		

6.3. Add SIP Entity – Communication Manager

Add Communication Manager as a SIP Entity. Navigate to **Home** → **Elements** → **Routing** → **SIP Entities**, click on **New** (no shown) and configure as follows:

- Type in a descriptive name in **Name** field
- Type in the IP address or FQDN of Communication Manager in **FQDN or IP Address** field.
- Set **Type** to **CM**
- Set **Location** to the location configured in **Section 6.2**

Click **Commit** to save changes.

Note: It is assumed that SIP Entity for Session Manager has already been configured.

SIP Entity Details

CommitCancel

General

* Name:

cm-tr1

* FQDN or IP Address:

10.64.10.67

Type:

CM

▼

Notes:

Adaptation:

▼

Location:

▼

Time Zone:

America/Fortaleza

▼

Override Port & Transport with DNS SRV:

☐

* SIP Timer B/F (in seconds):

4

Credential name:

Call Detail Recording:

none

▼

Loop Detection

Loop Detection Mode:

Off

▼

SIP Link Monitoring

SIP Link Monitoring:

Use Session Manager Configuration

▼

6.4. Add Adaptation

Navigate to **Home → Elements → Routing → Adaptation**, click **New** (not shown) and configure as follows:

- Type in a descriptive name in **Adaptation Name** field
- Type in **DigitConversionAdapter** in **New Module Name** field
- In the **Module Parameter** field type in the following:

iodstd=**domain** odstd=**wcs-ip-address** fromto=true osrcd=**sm-ip-address**

domain: Domain as configured in **Section 6.1**

wcs-ip-address: Wesley Clover Solutions IP Address

sm-ip-address: Session Manager IP Address

Adaptation Details

[Commit](#) [Cancel](#)

General

* **Adaptation name:**

Module name:

Module parameter:

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

[Add](#) [Remove](#)

0 Items Refresh		Filter: Enable							
<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes

Digit Conversion for Outgoing Calls from SM

[Add](#) [Remove](#)

0 Items Refresh		Filter: Enable							
<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes

6.5. Add SIP Entity – Wesley Clover Solutions IP PBX

Add Wesley Clover Solutions IP PBX as a SIP Entity. Navigate to **Home → Elements → Routing → SIP Entities**, click on **New** (no shown) and configure as follows:

- Type in a descriptive name in **Name** field
- Type in the IP address or FQDN of Wesley Clover IP PBX in **FQDN or IP Address** field
- Set **Type** to **SIP Trunk**
- Set **Adaptation** to the one configured in the previous section
- Set **Location** to the location configured in **Section 6.2**

Click **Commit** to save changes.

SIP Entity Details

Commit **Cancel**

General

* **Name:**

* **FQDN or IP Address:**

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV: ☐

* **SIP Timer B/F (in seconds):**

Credential name:

Call Detail Recording:

Loop Detection

Loop Detection Mode:

SIP Link Monitoring

SIP Link Monitoring:

6.6. Add Entity Link – Communication Manager

Navigate to **Home → Elements → Routing → Entity Links**, click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field
- Set **SIP Entity 1** to the name of Session Manager SIP Entity
- Set **SIP Entity 2** to Communication Manager SIP Entity configured in **Section 6.3**

Click **Commit** to save changes.

Entity Links Commit Cancel Help

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	
<input type="checkbox"/>	*Communication Ma	*asm-tr1	TLS	*5061	*cm-tr1	*5061	trusted	<input type="checkbox"/>	

< >

Select : All, None

6.7. Add Entity Link – Wesley Clover Solutions IP PBX

Navigate to **Home → Elements → Routing → Entity Links**, click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field
- Set **SIP Entity 1** to the name of Session Manager SIP Entity
- Set **Protocol** to **UDP**
- Set **SIP Entity 2** to Wesley Clover Solutions IP PBX SIP Entity configured in **Section 6.5**

Click **Commit** to save changes.

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	
<input type="checkbox"/>	*asm-tr1_wc-tr1_506	*asm-tr1	UDP	*5060	*wc-tr1	*5060	trusted	

< >

Select : All, None

6.8. Add Time Ranges

Navigate to **Home → Elements → Routing → Time Ranges**, click on **New** (now shown) and configure as follows:

- Type in a descriptive name in **Name** field

Click **Commit** to save changes.

Time Ranges

CommitCancel

1 ItemRefresh

Filter: Enable

Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
*TimeRange	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	*00:00	*23:59	

6.9. Add Routing Policy

Navigate to **Home → Elements → Routing → Routing Policies**, click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field
- Under **SIP Entity as Destination**, click on **Select** (not shown):
 - Select Wesley Clover Solutions IP PBX SIP entity added in **Section 6.5**
- Under **Time of Day**, click on **Add** (not shown):
 - Select time range added in previous step

Click **Commit** to save changes.

Routing Policy Details

[Commit](#) [Cancel](#)

General

* **Name:**

Disabled: ☐

* **Retries:**

Notes:

SIP Entity as Destination

[Select](#)

Name	FQDN or IP Address	Type	Notes
wc-tr1	10.64.10.170	SIP Trunk	

Time of Day

[Add](#) [Remove](#) [View Gaps/Overlaps](#)

1 Item Refresh		Filter: Enable										
<input type="checkbox"/>	Ranking ▲	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	
Select : All, None												

6.10. Add Dial Patterns

Navigate to **Home → Elements → Routing → Dial Patterns**, click on **New** (not shown) and configure as follows:

Under **General**:

- Set **Pattern** to prefix of dialed number
- Set **Min** to minimum length of dialed number
- Set **Max** to maximum length of dialed number

Under **Originating Locations and Routing Policies**:

- Click **Add** and select originating location and Wesley Clover Solutions IP PBX routing policy as configured in **Section 6.9**

Click **Commit** to save changes.

Dial Pattern Details

[Commit](#) [Cancel](#)

General

* **Pattern:**


* **Min:**

* **Max:**

Emergency Call: ☐

Emergency Priority:

Emergency Type:

SIP Domain: 

Notes:

Originating Locations and Routing Policies

[Add](#) [Remove](#)

1 Item Refresh		Filter: Enable					
<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Test Room 1		wc-tr1	0	<input type="checkbox"/>	wc-tr1	
Select : All, None							

7. 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 the Wesley Clover Solutions IP PBX and Avaya Aura® Communication Manager.

7.1. Assumptions

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

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.

7.2. 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**:

- 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

7.3. 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 7.2**
- 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

7.4. 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 Session Manager in **FQDN or IP Address** field
- Set **Zone** of 1
- Select **SIP Peer** selection box

Network Elements	
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

Save Cancel

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 Session Manager in the **FQDN or IP Address** field

Network Elements	
Name	AvayaPrxy
Type	Outbound Proxy
FQDN or IP Address	10.64.10.62
Local	False
Version	
Zone	1
ARID	
Outbound Proxy Specific	
Outbound Proxy Transport Type	default
Outbound Proxy Port	5060
<div>Save Cancel</div>	

7.5. 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 7.4**
- Leave the **Registration User Name** field blank
- Enter the **Maximum Simultaneous Calls**. This is the number of SIP trunks to be used between Session Manager 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 7.4**
- In the Trunk Service field enter the **Trunk Service Number** created in **Section 7.3**

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, 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	

7.6. 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. In the following example an extension range of 4000-5002 is used:

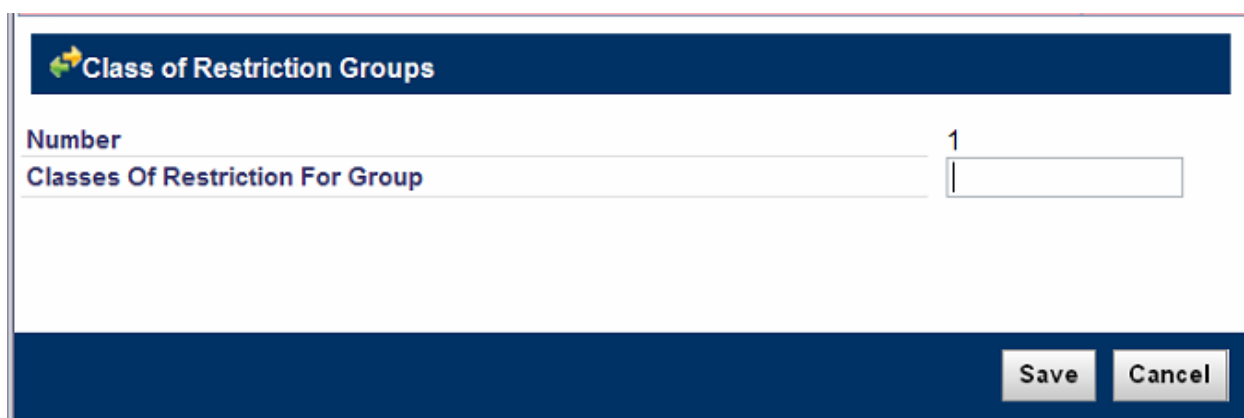
- Click Add (not shown)
- Enter extension ranges in the **Incoming DID Range** field
- Select the **SIP Peer Profile Label** created in **Section 7.5** in the drop down box
- Add a meaningful comment in the **Comment** field

SIP Peer Profile Assignment by Incoming DID	
Incoming DID Range	4000-5002
SIP Peer Profile Label	Avaya ▼
Comment	Avaya SIP

7.7. Program the Class Of Restriction Group Form

Navigate to **System Properties → System Feature Settings → Class of Restriction Groups** (not shown)

Verify that the class 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.



Class of Restriction Groups

Number 1

Classes Of Restriction For Group

Save Cancel

7.8. 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' which is empty, and 'Final Tone Plan/Information Marker' which is empty. At the bottom right, there are 'Save' and 'Cancel' buttons.

7.9. 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 7.5**
- Enter the **COR Group Number** created in **Section 7.7**
- Enter the **Digit Modification Number** created in **Section 7.8**

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' which is empty, 'SIP Peer Profile' with a dropdown menu showing 'Avaya', 'PBX Number / Cluster Element ID' which is empty, 'COR Group Number' with the value '1', 'Digit Modification Number' with the value '1', 'Digits Before Outpulsing' with a dropdown menu, 'Route Type' with a dropdown menu, and 'Compression' with a dropdown menu showing 'Off'. At the bottom right, there are 'Save' and 'Cancel' buttons.

7.10.Program ARS Digits Dialed Form

Navigate to **Call Routing → Automatic Route Selection (ARS) → ARS Digits Dialed**

In this example the Avaya extension 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 7.9**

Add Range Programming - ARS Digits Dialed Help
This form allows you to add one or more records.

1. Enter the number of records to add:

2. Define the Add Range Programming Pattern:

Field Name	Value to Add	Increment by
Digits Dialed	<input type="text" value="2"/>	<input type="text"/>
Number of Digits to Follow	<input type="text" value="4"/>	<input type="text"/>
Termination Type	<input type="text" value="Route"/>	<input type="text"/>
Termination Number	<input type="text" value="2"/>	<input type="text"/>

Preview Save Cancel

7.11.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.

Shared System Options

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

8. Verification Steps

8.1. Avaya Aura® Session Manager

From the System Manager web page, navigate to **Session Manager → System Status → SIP Entity Monitoring**. Under the **All Monitoring SIP Entities**, select Wesley Clover Solutions IP PBX SIP entity that was configured in this document (not shown).

Ensure that **Conn. Status** is **UP**, and **Reason Code** is **200 OK**. This will verify that the connection between Session Manager and Wesley Clover Solutions IP PBX is successful.

1 Items Refresh Filter: Enable									
	Session Manager	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status	
<input type="radio"/>	asm-tr1	10.64.101.9	5060	TCP	FALSE	UP	200 OK	UP	

8.2. Welsey 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.

9. Conclusion

Wesley Clover Solutions Trading Platform was able to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All executed test cases passed.

10. Additional References

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

- [1] Administering Avaya Aura® Communication Manager, Release 6.3, Document 03-3005089, Issue 7.0, December 2012
- [2] Administering Avaya Aura® Session Manager, Release 6.3, Document 03-603324, July 2012

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

©2013 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.