Avaya Solution & Interoperability Test Lab

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.

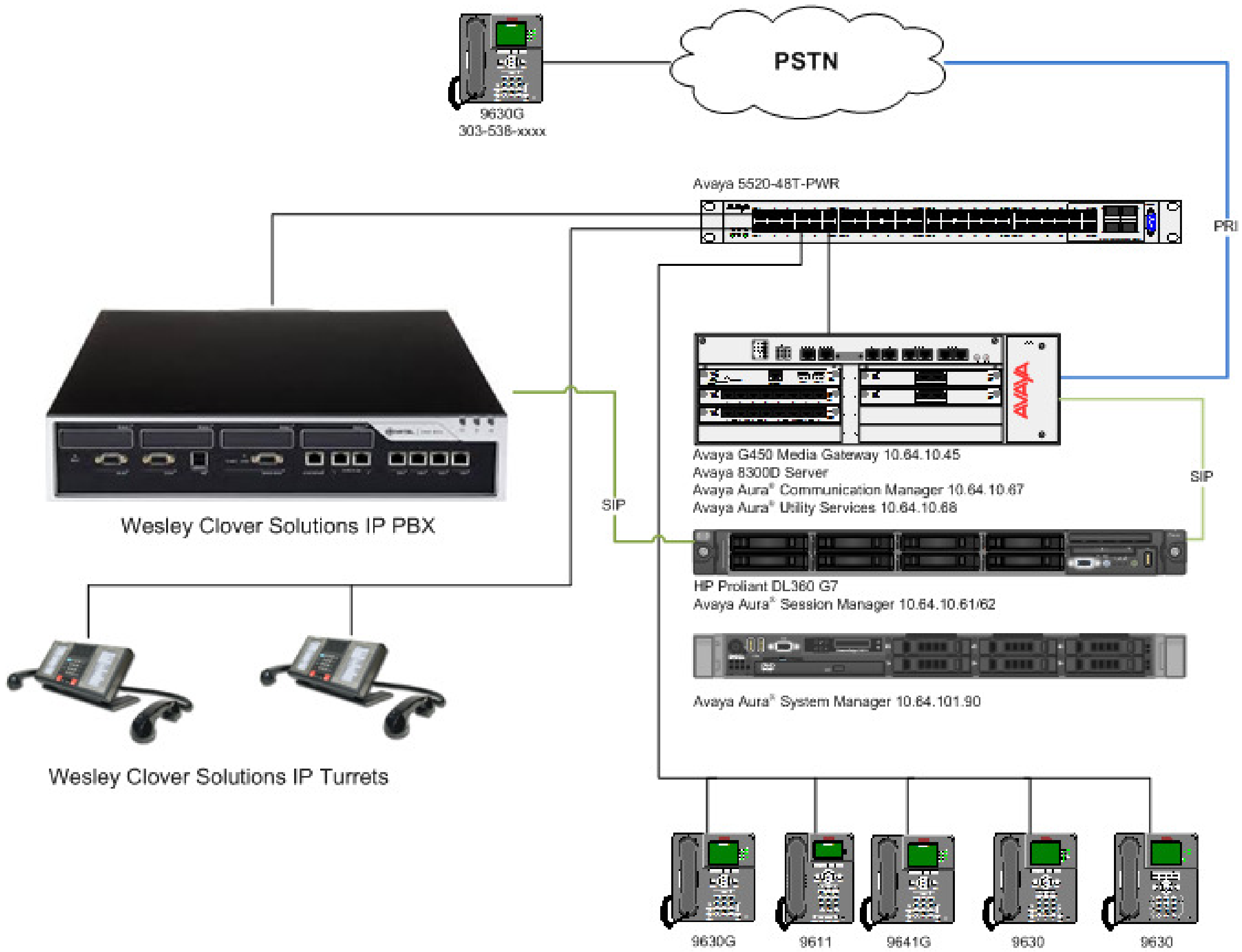


Figure 1: Reference Configuration

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 Avaya 96x0 Series H.323 Phones	6.2.4 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      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP DESKPHONES                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 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        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  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
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing 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
 Dial Access? n                                 Night Service:
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
                                                UUI 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
                             Dgts                                      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.

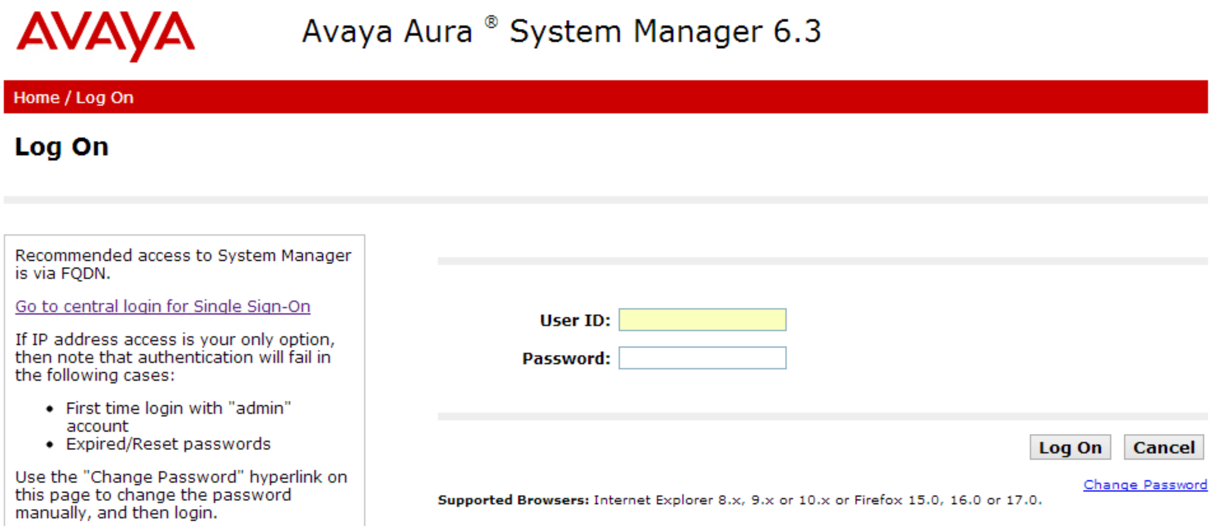
```
change aar analysis 4                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    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

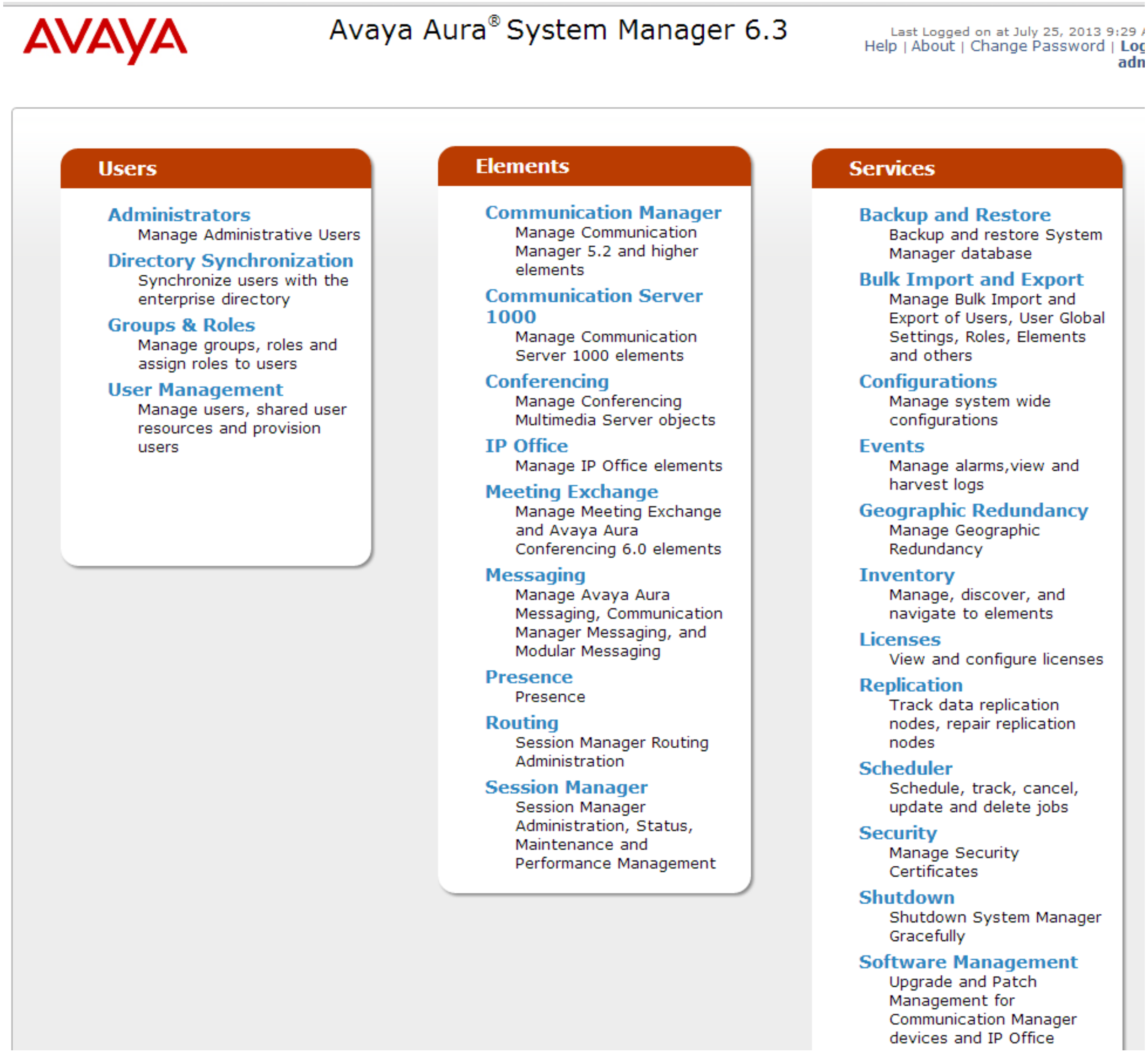
Administration of Avaya Stations/Extensions in Communication Manager and Session Manager is not shown in this document. Please refer to document [1] and/or [2] in reference section of this document.

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.



Log in using appropriate credentials.

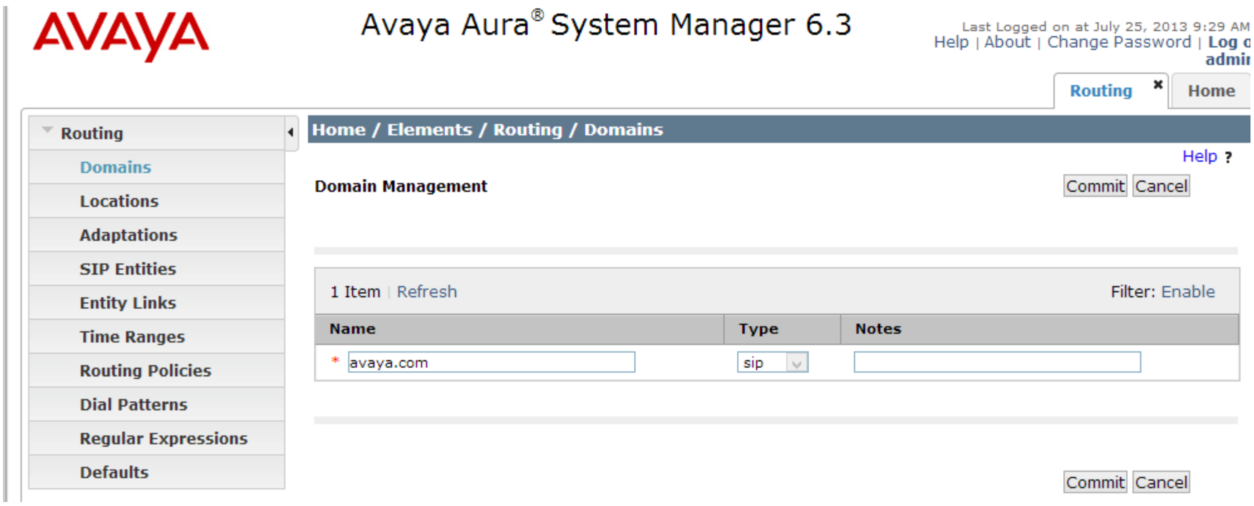


6.1. Add SIP Domain

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.



6.2. Add Location

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

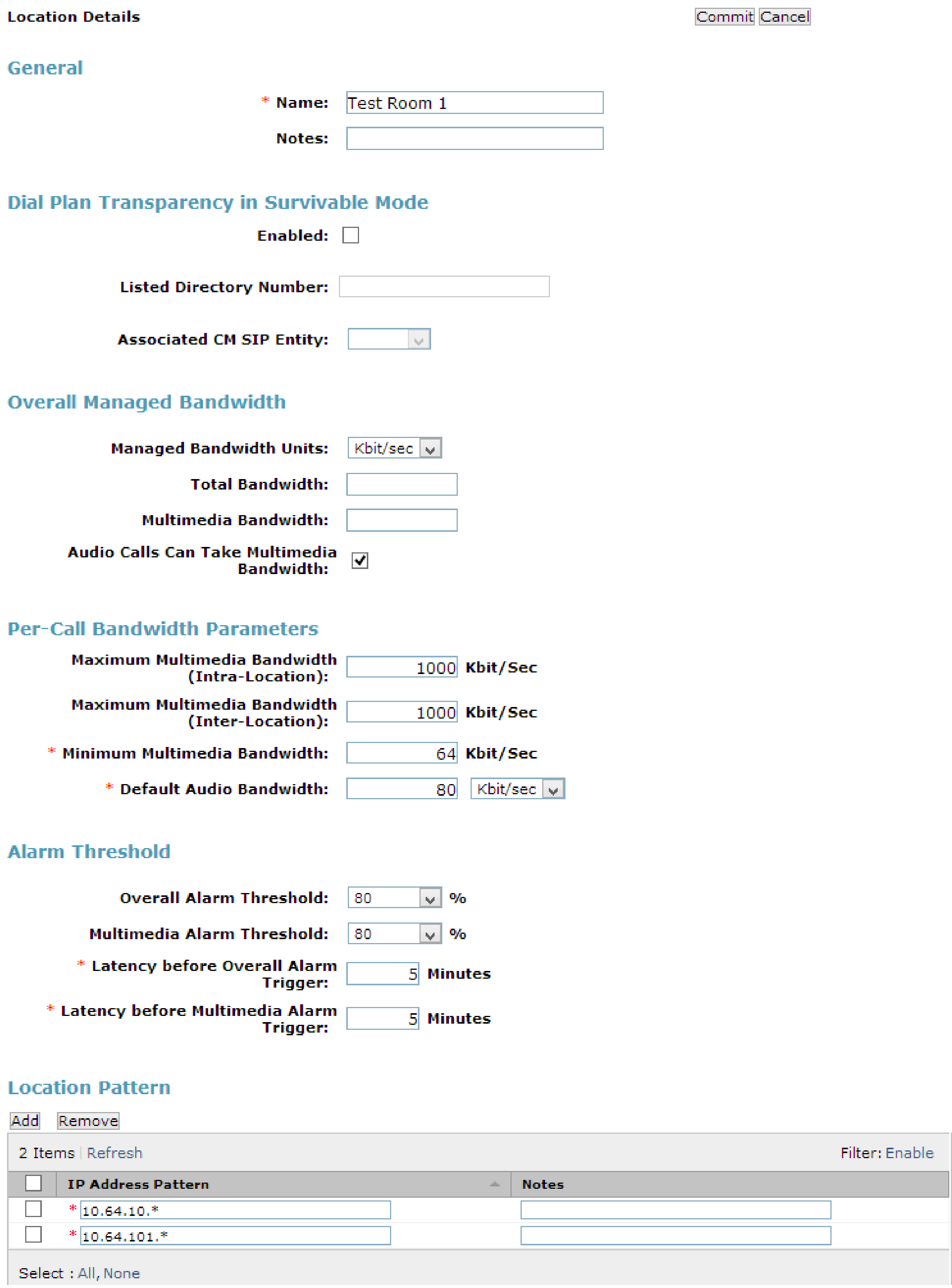
Under **General**:

- 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:
Test Room 1
Notes:
Dial Plan Transparency in Survivable Mode
Enabled:
Listed Directory Number:
Associated CM SIP Entity:
Overall Managed Bandwidth
Managed Bandwidth Units:
Kbit/sec
Total Bandwidth:
Multimedia Bandwidth:
Audio Calls Can Take Multimedia Bandwidth:
Per-Call Bandwidth Parameters
Maximum Multimedia Bandwidth (Intra-Location):
1000
Kbit/Sec
Maximum Multimedia Bandwidth (Inter-Location):
1000
Kbit/Sec
* Minimum Multimedia Bandwidth:
64
Kbit/Sec
* Default Audio Bandwidth:
80
Kbit/sec
Alarm Threshold
Overall Alarm Threshold:
80
%
Multimedia Alarm Threshold:
80
%
* Latency before Overall Alarm Trigger:
5
Minutes
* Latency before Multimedia Alarm Trigger:
5
Minutes
Location Pattern
Add
Remove
2 Items | Refresh
Filter: Enable
IP Address Pattern
Notes
* 10.64.10.*
* 10.64.101.*
Select : All, None

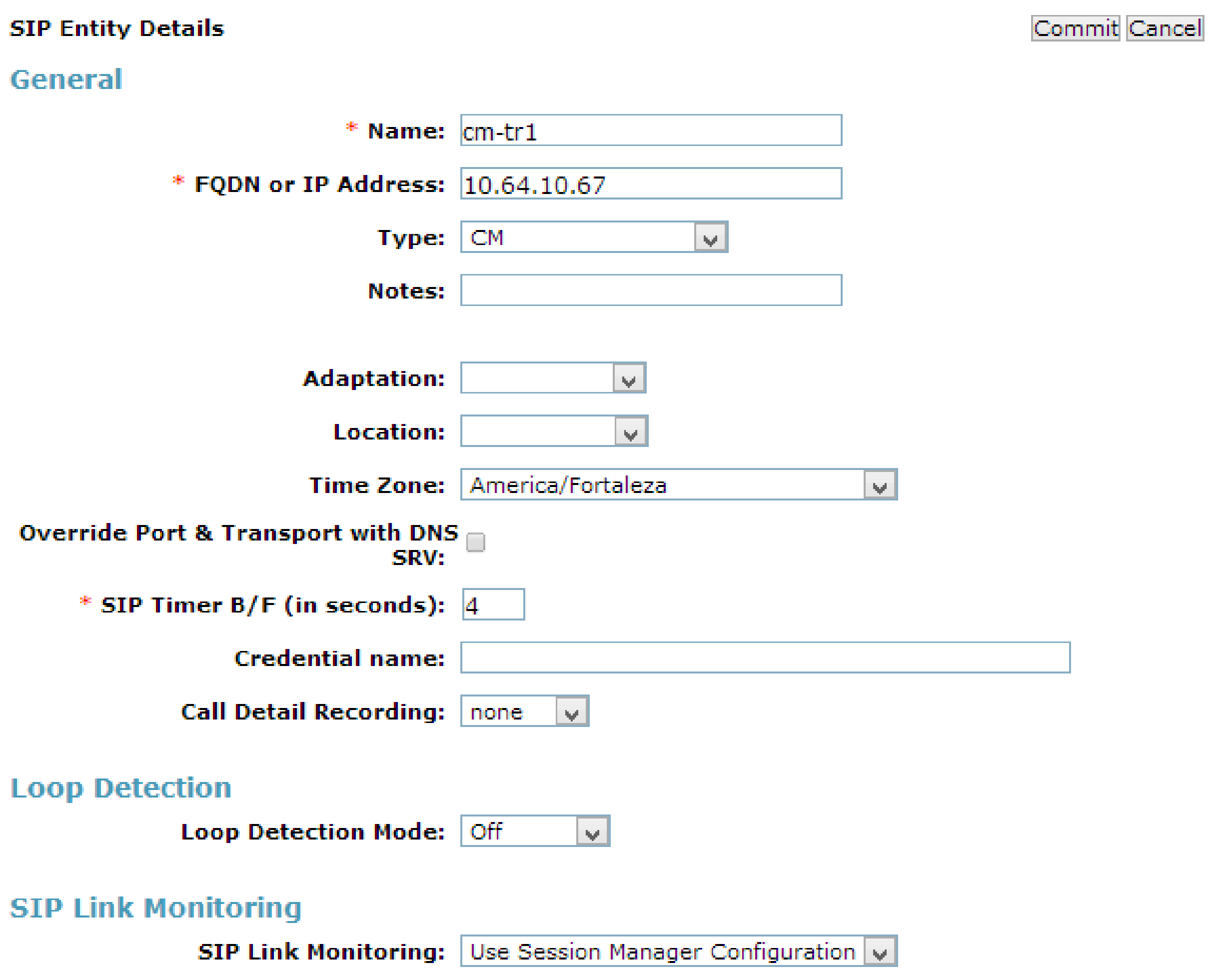
6.3. Add SIP Entity – Communication Manager

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.



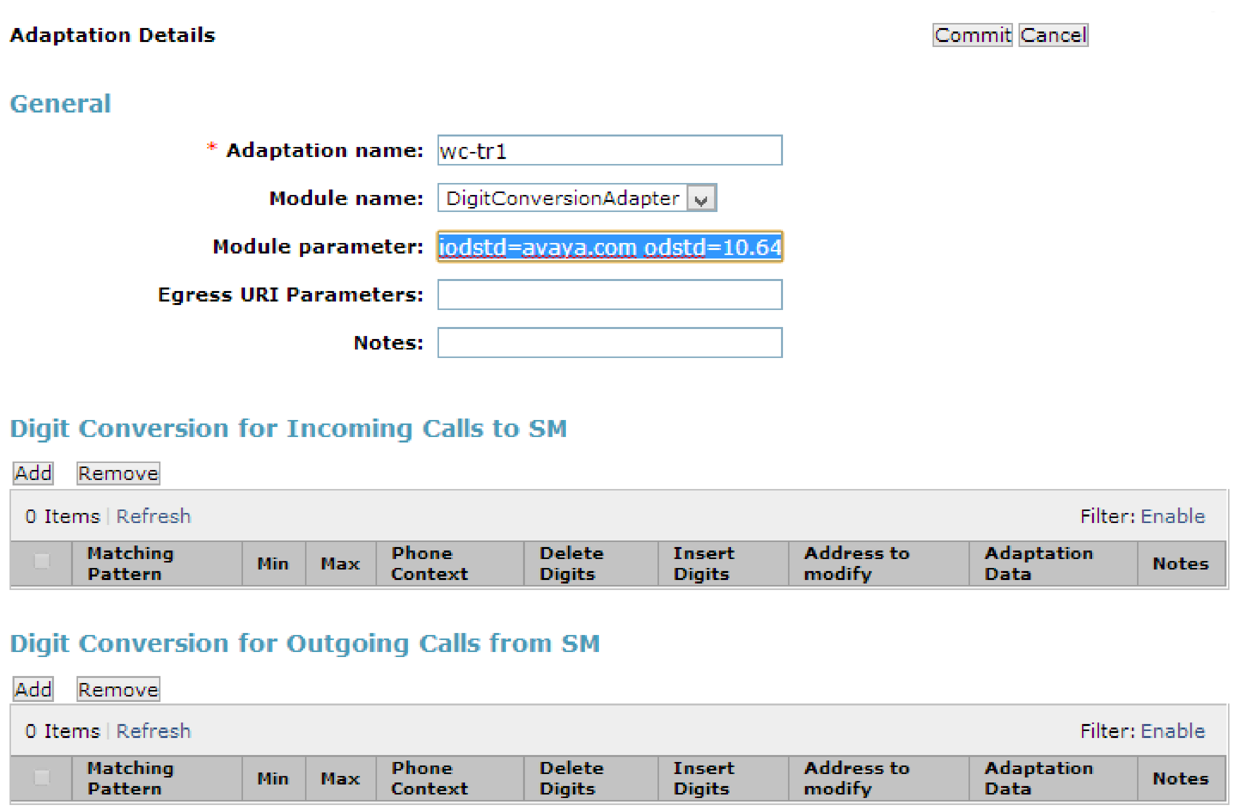
6.4. Add Adaptation

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

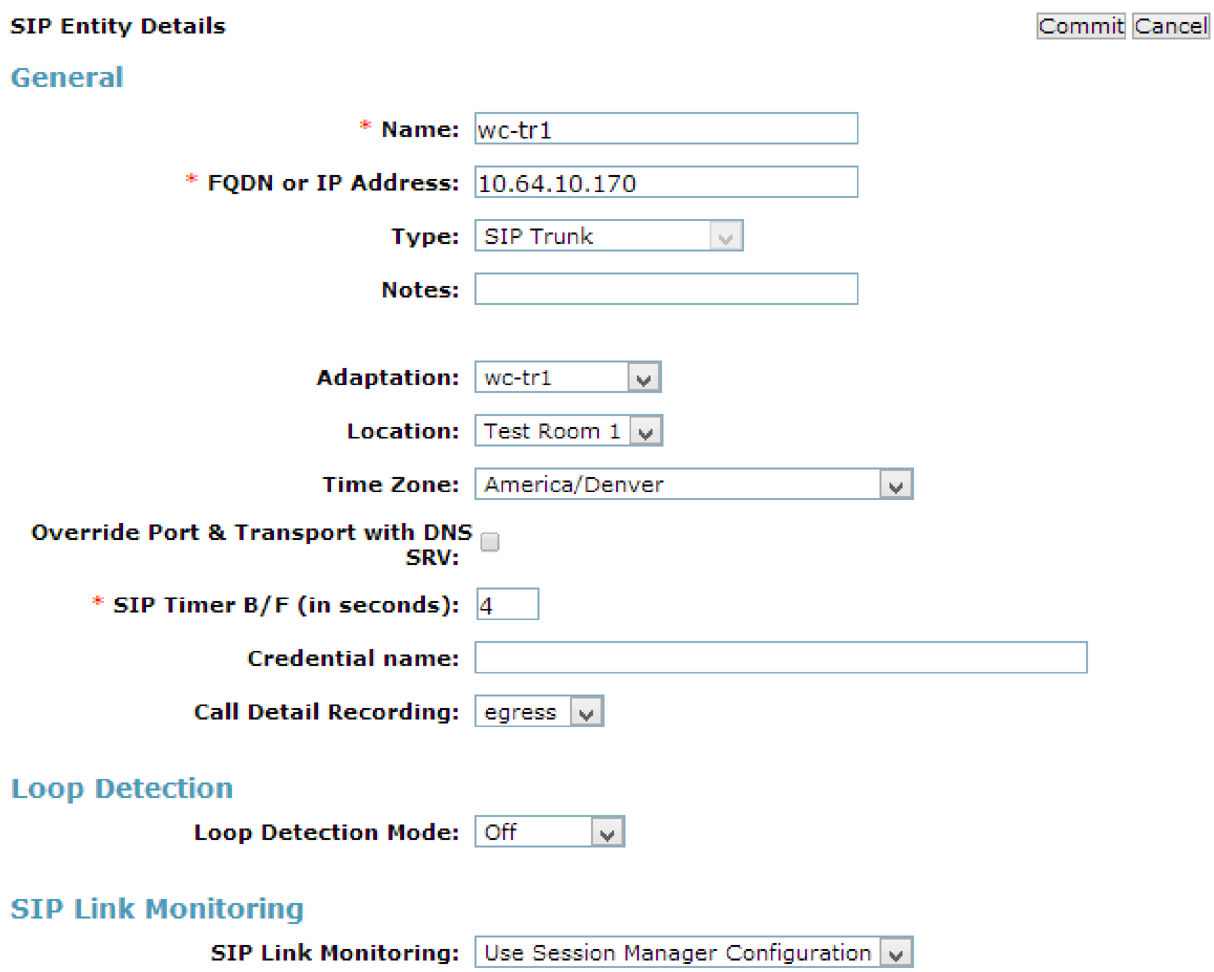


6.5. Add SIP Entity – Wesley Clover Solutions IP PBX

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.

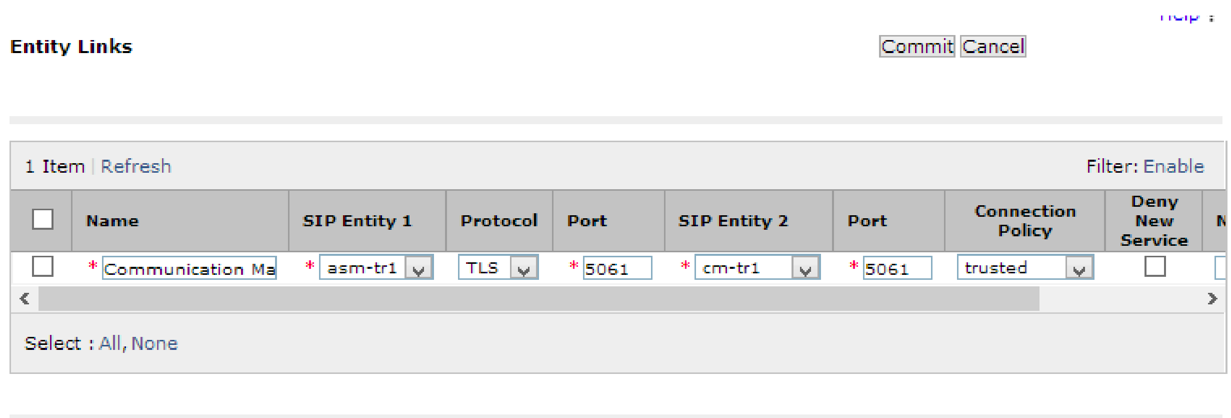


6.6. Add Entity Link – Communication Manager

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.

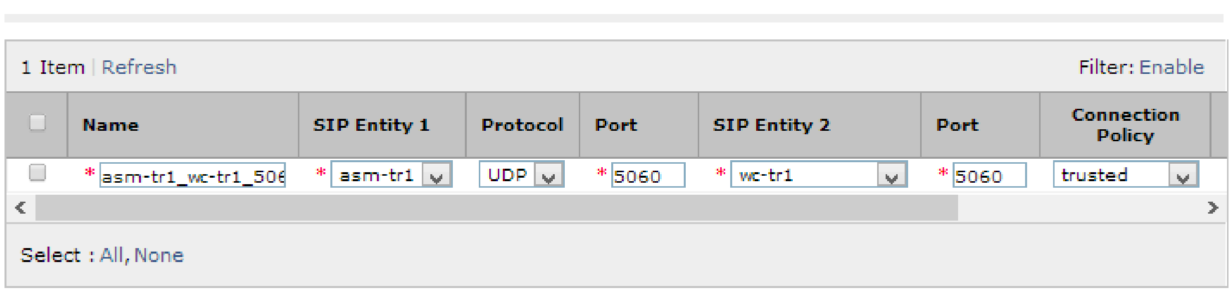


6.7. Add Entity Link – Wesley Clover Solutions IP PBX

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.

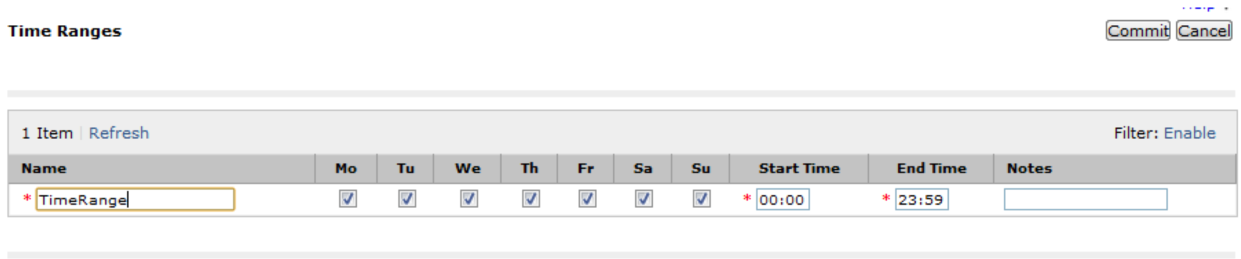


6.8. Add Time Ranges

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.

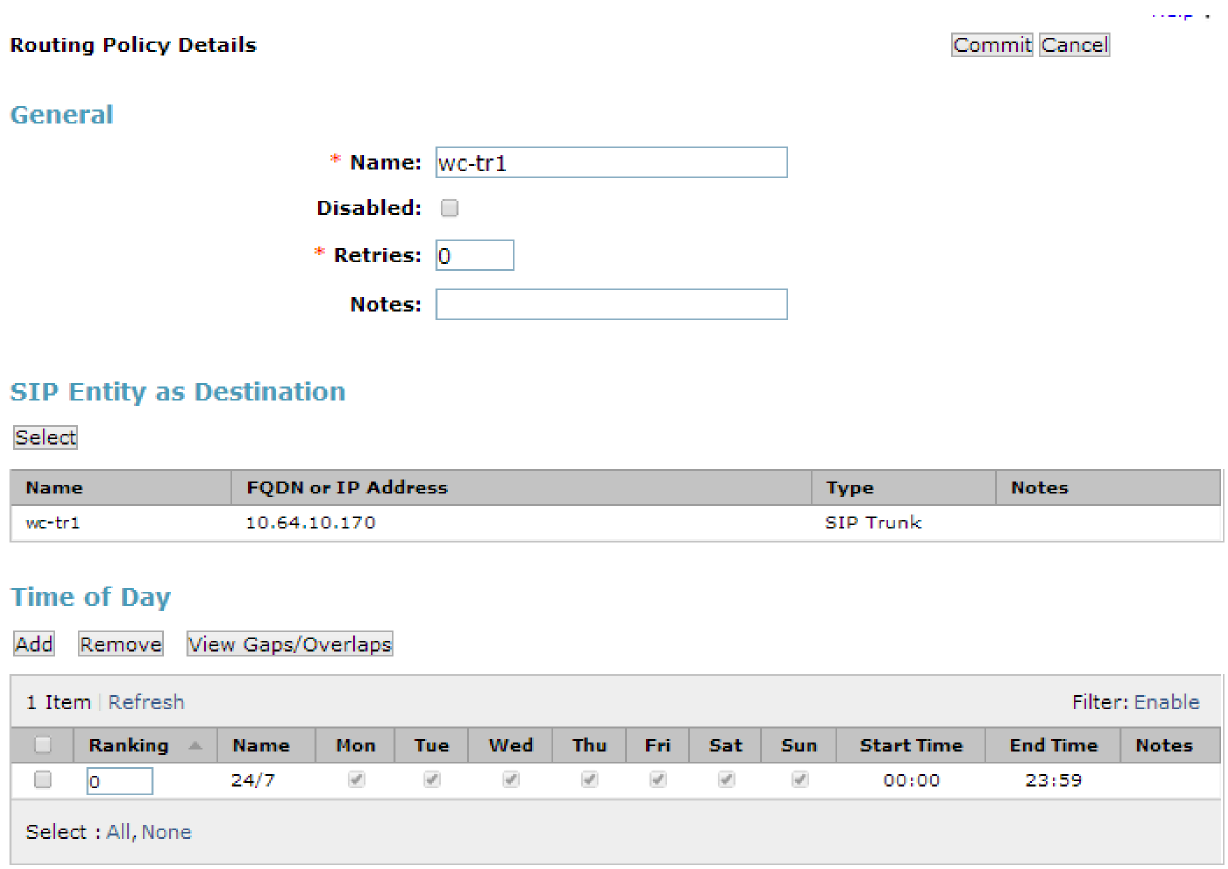


6.9. Add Routing Policy

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):
  - o Select Wesley Clover Solutions IP PBX SIP entity added in **Section 6.5**
- Under **Time of Day**, click on **Add** (not shown):
  - o Select time range added in previous step

Click **Commit** to save changes.



6.10. Add Dial Patterns

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

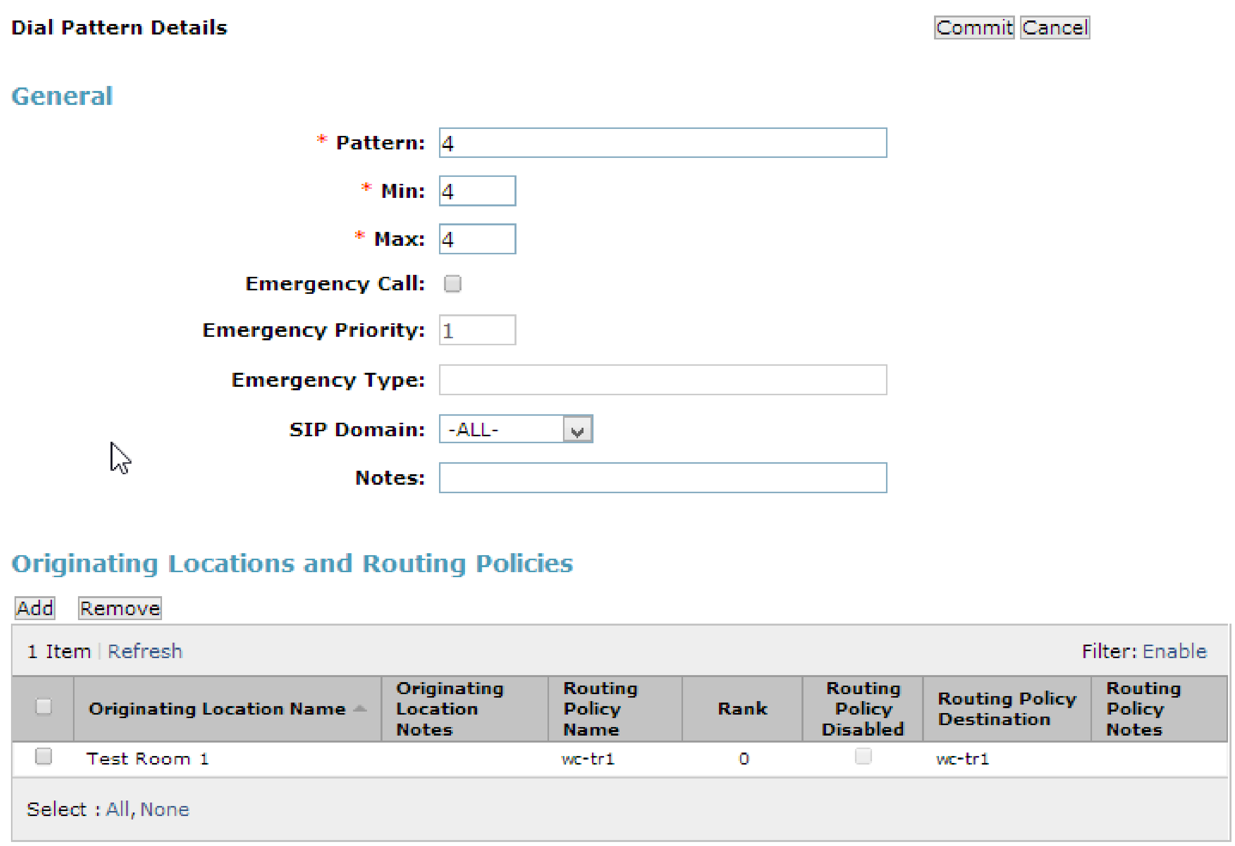
Under **General:**

- 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.



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)

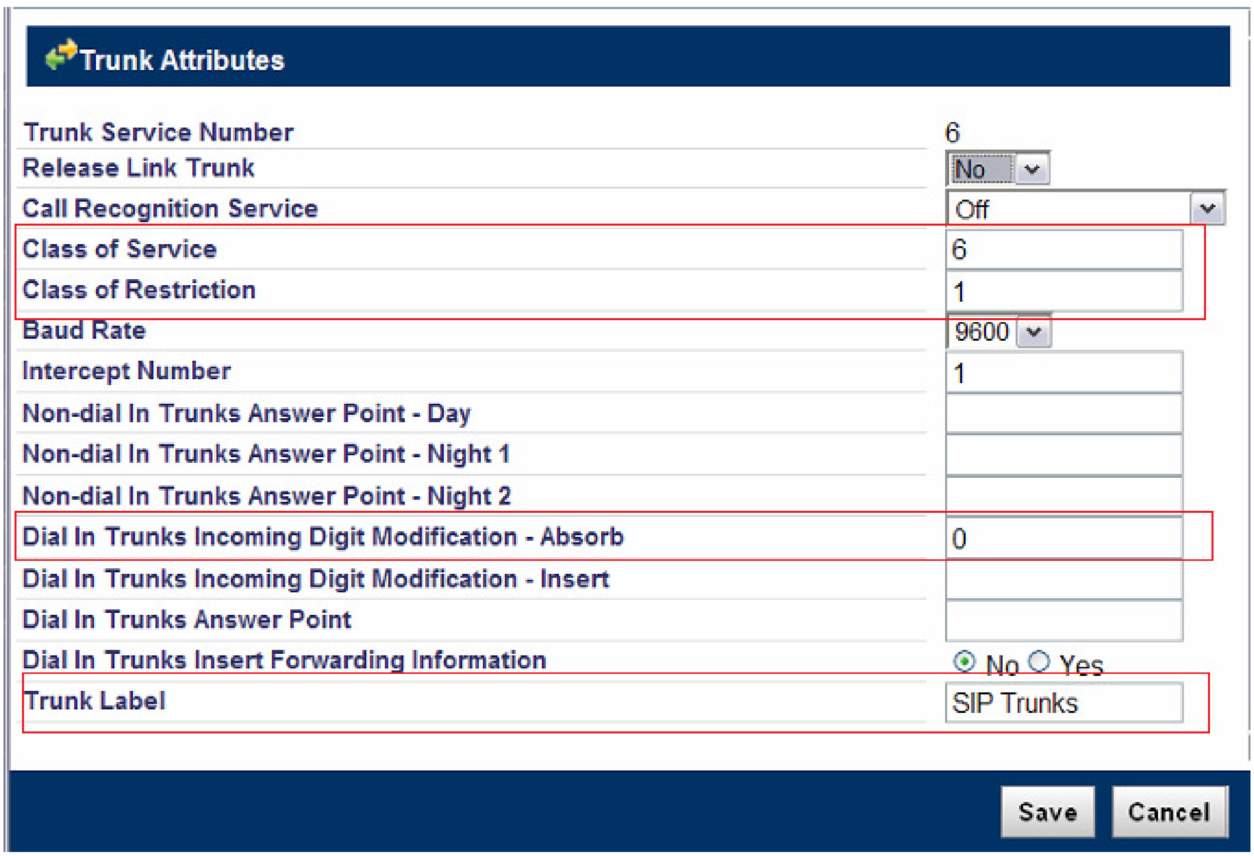
Program a unique COS, in this case **6** is used (not shown) and set the following trunk options to **Yes**:

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

7.3. Program Trunk Attributes Form

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

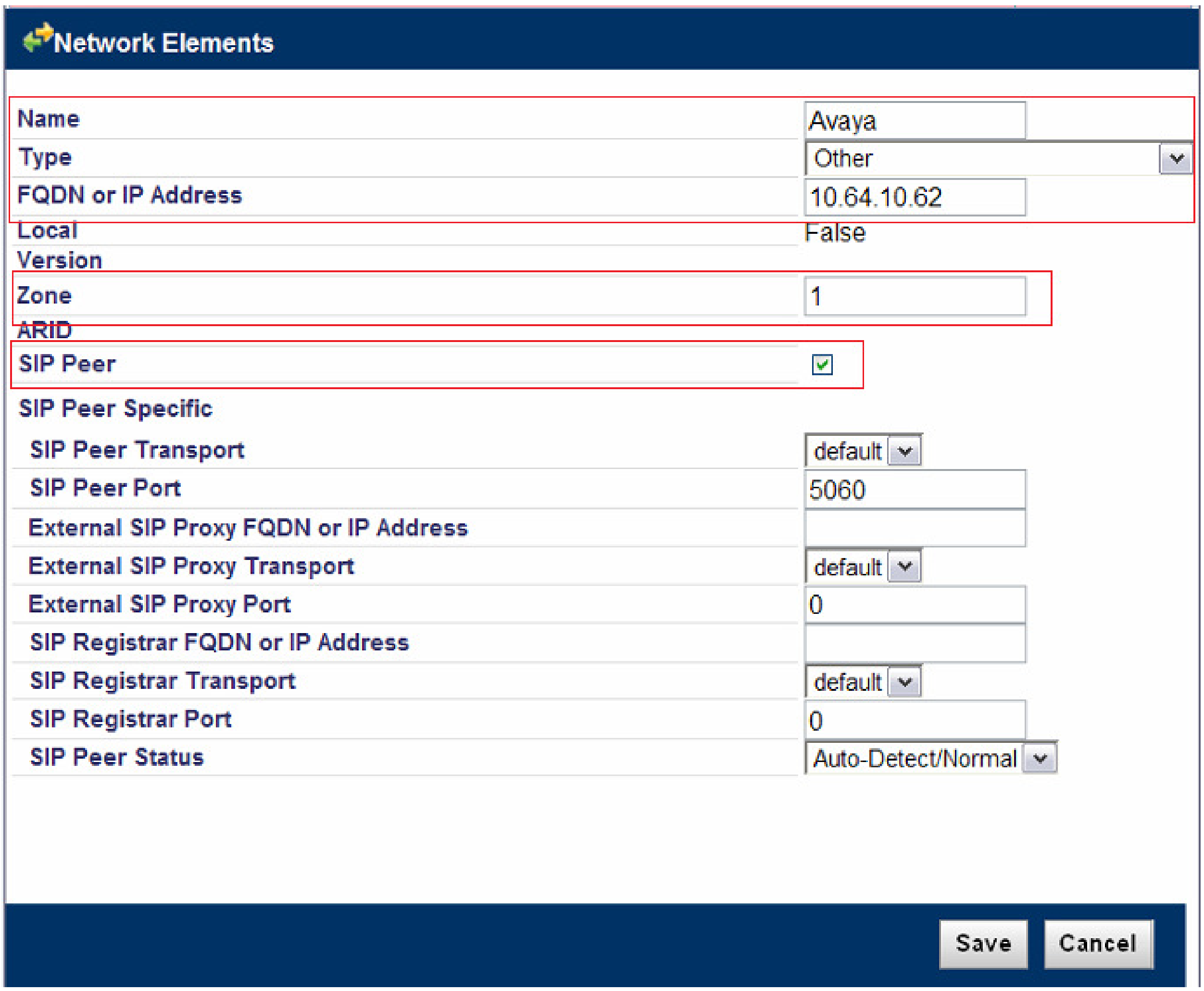
- Set the **Class of Service** to the COS assigned in **Section 7.2**
- Set **Class of Restriction** to **1**
- Set the **Dial-In Trunk Incoming Digit Modification – Absorb** to **0**
- Add a **Trunk Label**



7.4. Program the Network Elements Form

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

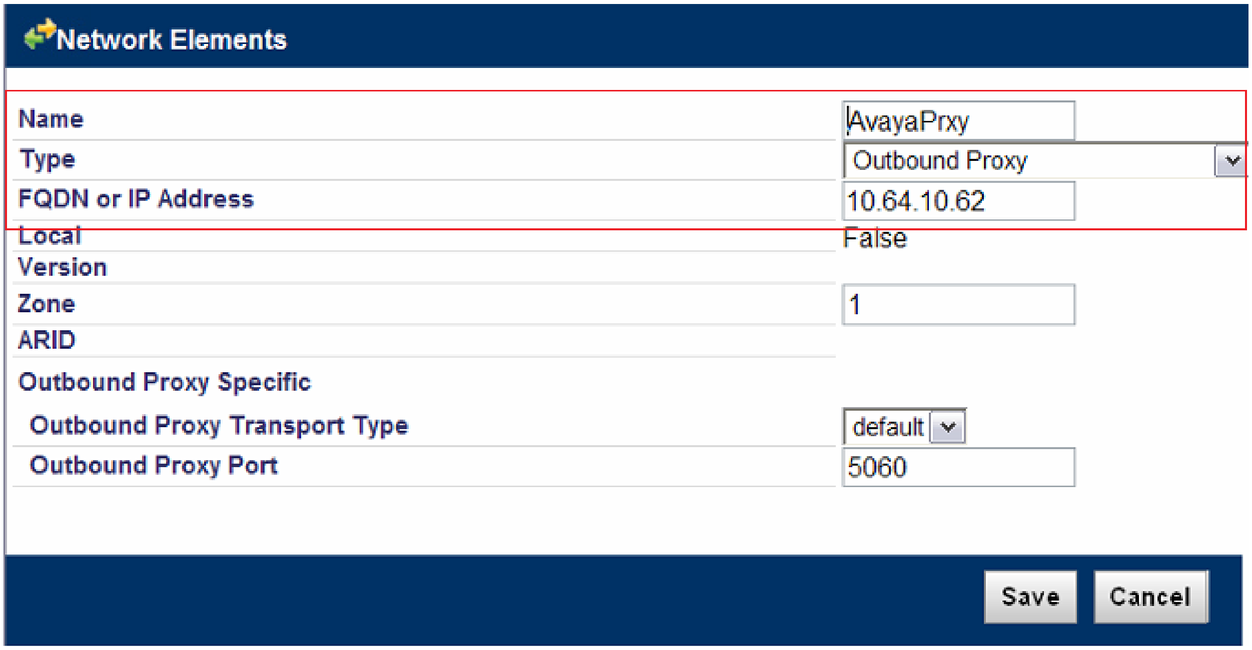
- Enter a name in the **Name** Field. For example, “Avaya”
- Select **Other** in the **Type** drop down box
- Enter the IP address of Session Manager in **FQDN or IP Address** field
- Set **Zone** of 1
- Select **SIP Peer** selection box



Program the Network Elements Form (Continued)

Configure a second Network Element for the proxy.

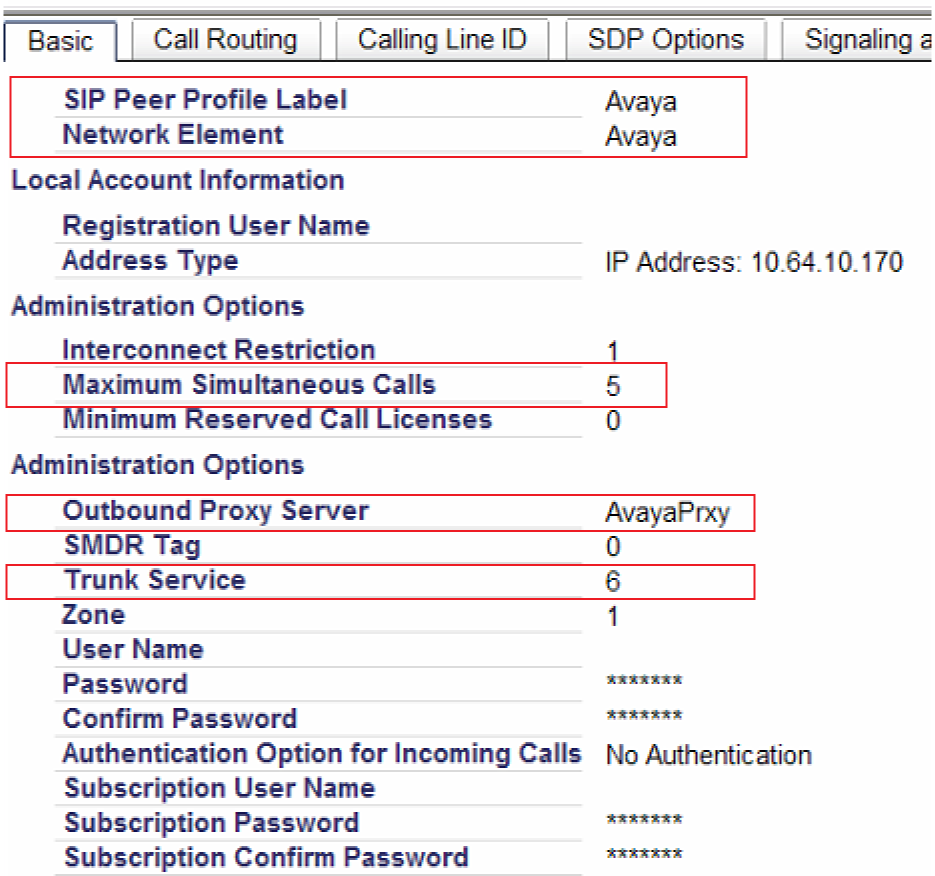
- Enter a meaningful name in the **Name** field. For example, "AvayaPrxy"
- Select **Outbound Proxy** in the **Type** drop down
- Enter the IP address of Session Manager in the **FQDN or IP Address** field



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**



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

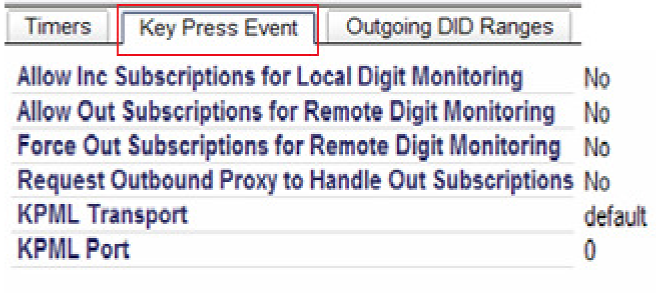
Program SIP Peer Profile Form (Continued)

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

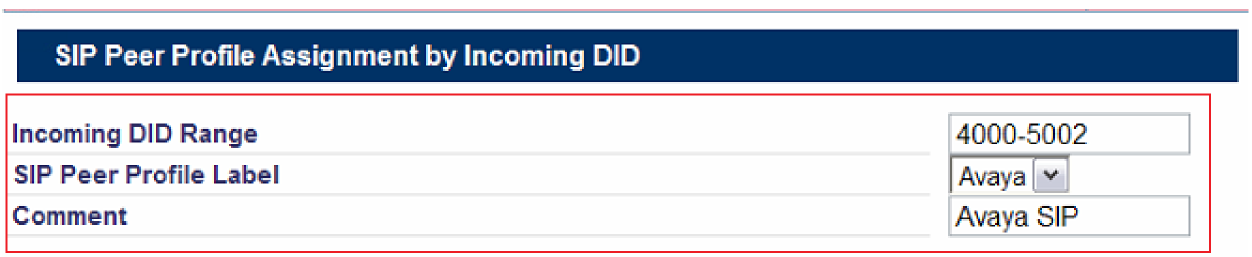
Keep-Alive (OPTIONS) Period	120
Registration Period	3600
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	90
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0



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:

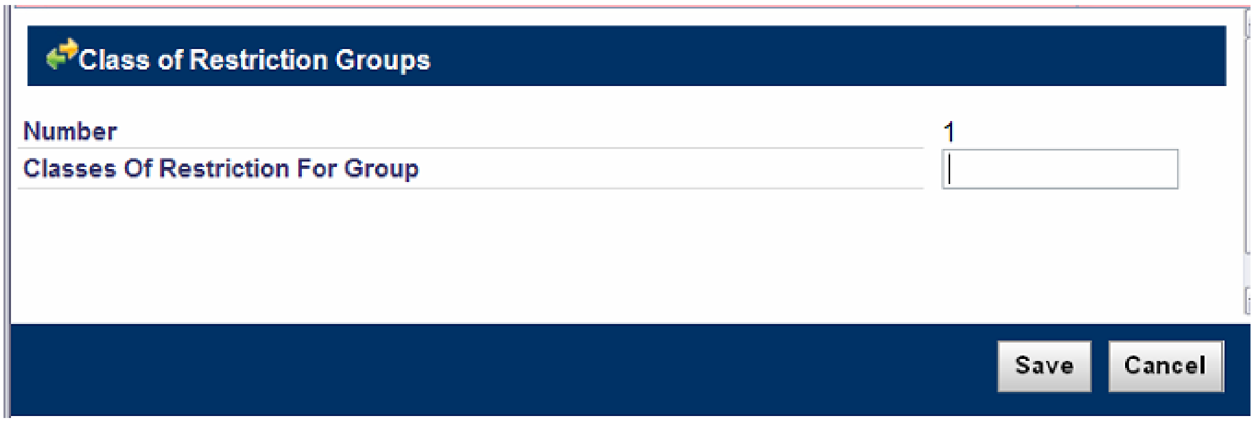
- Click Add (not shown)
- Enter extension ranges in the **Incoming DID Range** field
- Select the **SIP Peer Profile Label** created in **Section 7.5** in the drop down box
- Add a meaningful comment in the **Comment** field



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.



7.8. Program ARS Digit Modification Plans Form

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

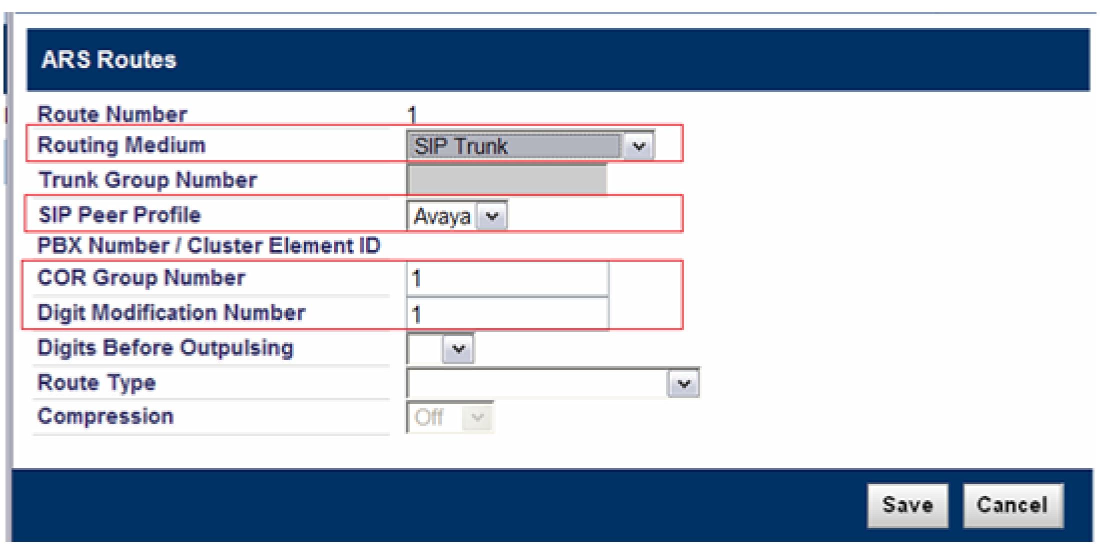
In this example **Digit Modification Number** of **1** is used. Set the **Number of Digits to Absorb** to **0**.

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.

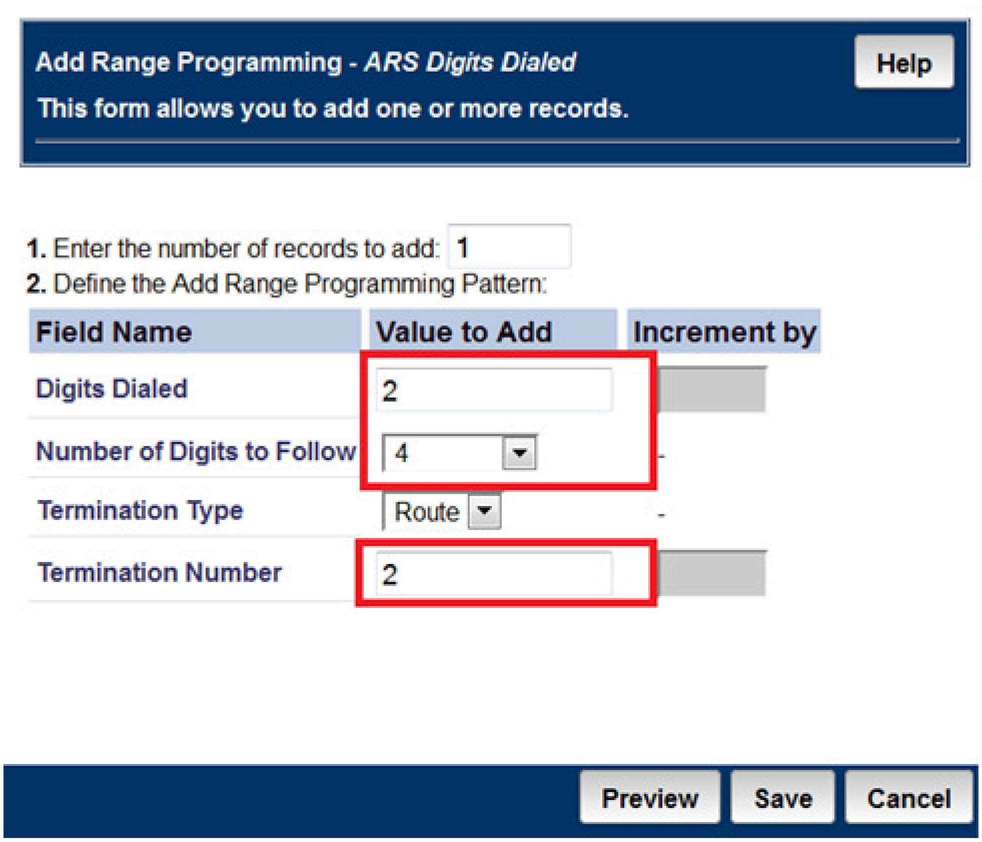
- In the **Routing Medium** drop down box select **SIP Trunk**
- In the **SIP Peer Profile** select the peer created in **Section 7.5**
- Enter the **COR Group Number** created in **Section 7.7**
- Enter the **Digit Modification Number** created in **Section 7.8**



7.10.Program ARS Digits Dialed Form

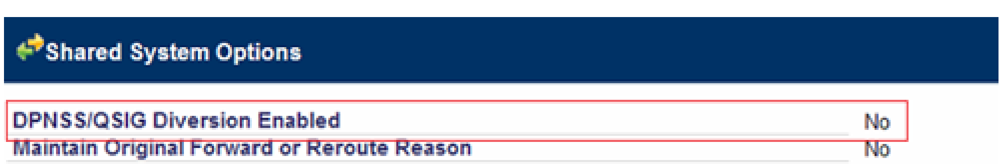
Navigate to **Call Routing → Automatic Route Selection (ARS) → ARS Digits Dialed**
In this example the Avaya extension are 5 digits in length and begin with a 2.

- 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**



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.



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
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

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.