Interactive Intelligence Customer Interaction Center 4.0

PBX Configuration Note:
Avaya Aura with CIC using SIP

Technical Reference

By Interactive Intelligence, Inc.

Customer Interaction Center®

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Content

This document describes the configuration required to setup Avaya Aura to interoperate with Interactive Intelligence Customer Interaction Center 4.0 (CIC) using SIP.

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Interaction Center Platform Statement

This document describes Interaction Center (IC) features that may not be available in your IC product. Several products are based on the IC platform, and depending on your product and version, some features may not be available.

Table of Contents

Who should read this document?

This document is intended for Systems Integrators with significant telephony knowledge.

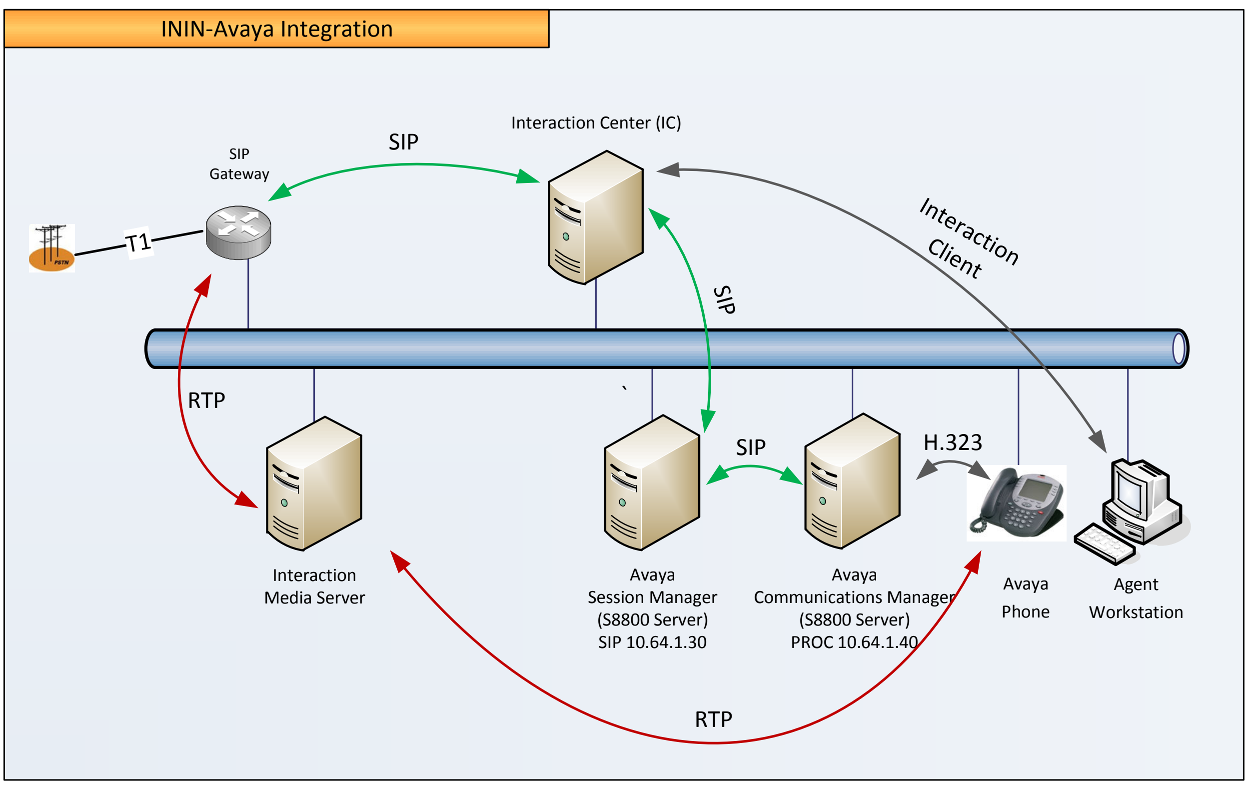
Technical Support

The information contained within this document has been provided by Interactive Intelligence, its partners or equipment manufacturers and is provided AS IS. This document contains information about how to modify the configuration of your PBX. Improper configuration may result in the loss of service of the PBX. Interactive Intelligence is unable to provide support or assistance with the configuration or troubleshooting of components described within. Interactive Intelligence recommends readers to engage the service of an Interactive Intelligence Certified Engineer or the manufacturers of the equipment(s) described within to assist with the planning and deployment of Customer Interaction Center.

Known Issues

None.

Sample Configuration



This above diagram is a sample configuration which enables Avaya Aura® Session Manager, Avaya Aura® Communication Manager, Interactive Intelligence Customer Interaction Center (CIC), and Interactive Intelligence Media Server to interoperate via SIP. The solution provided will allow calls to be delivered via a SIP tie line, which allows for intercom calls, delivery of calls into the CIC powered call center either directly from the PSTN or via the Avaya SIP tie line, and then delivery of call center calls to the agent using the Avaya phone as the voice path, also leveraging the SIP tie line.

Interoperability Supported:

- PSTN calls delivered via CIC Server SIP Gateway or SIP Trunking to an Avaya IP telephony solution
- PSTN calls sent via an Avaya Media Server or SIP Trunking (not shown).
- Calling with various Avaya telephone models including IP/SIP models as well as traditional analog and digital TDM phones
- G.711 / G.729 support
- DTMF Tone Support
- Codec negotiation
- Dial plan processing to deliver calls appropriately between CIC Call Center and Avaya Enterprise
- Telephony supplementary features, such as Hold, Call transfer, Conference Calling and Call Forwarding, powered by CIC Interaction Client
- Voicemail Delivery via TUI
- Direct IP-to-IP Media allowing for dynamic audio between Avaya phones and Interaction Media Server
- Calls routed in from PSTN through CIC Call Center to
- Calls routed in from PSTN through Avaya solution, to CIC Call Center, then back to Avaya for connection call to Agent Phone.
- Redundancy with the addition of Interaction SIP Proxy (not shown) to deliver calls to redundant CIC servers

Chapter 1: General Information

Components

PBX or IP-PBX

PBX Vendor	Avaya
Solution	Aura
Software Version	SM 6.0 and 6.1
Telephony Signaling	SIP
Additional Notes	None

Interactive Intelligence Customer Interaction Center

Software Version	4.0 SU 1

Prerequisites

PBX Prerequisites

Session Manager used as primary SIP route. CIC SIP Proxy can be used if desired. Configuration can be provided at testlab.inin.com.

Summary

This document provides for configuration of a SIP tie line between the CIC server and the Aura solution, which can then be leveraged for delivery of calls from one solution to the other, including using Avaya endpoints as remote dial voice paths for CIC applications, typically provisioned so that agents using Avaya hardware can receive CIC routed ACD calls and applications.

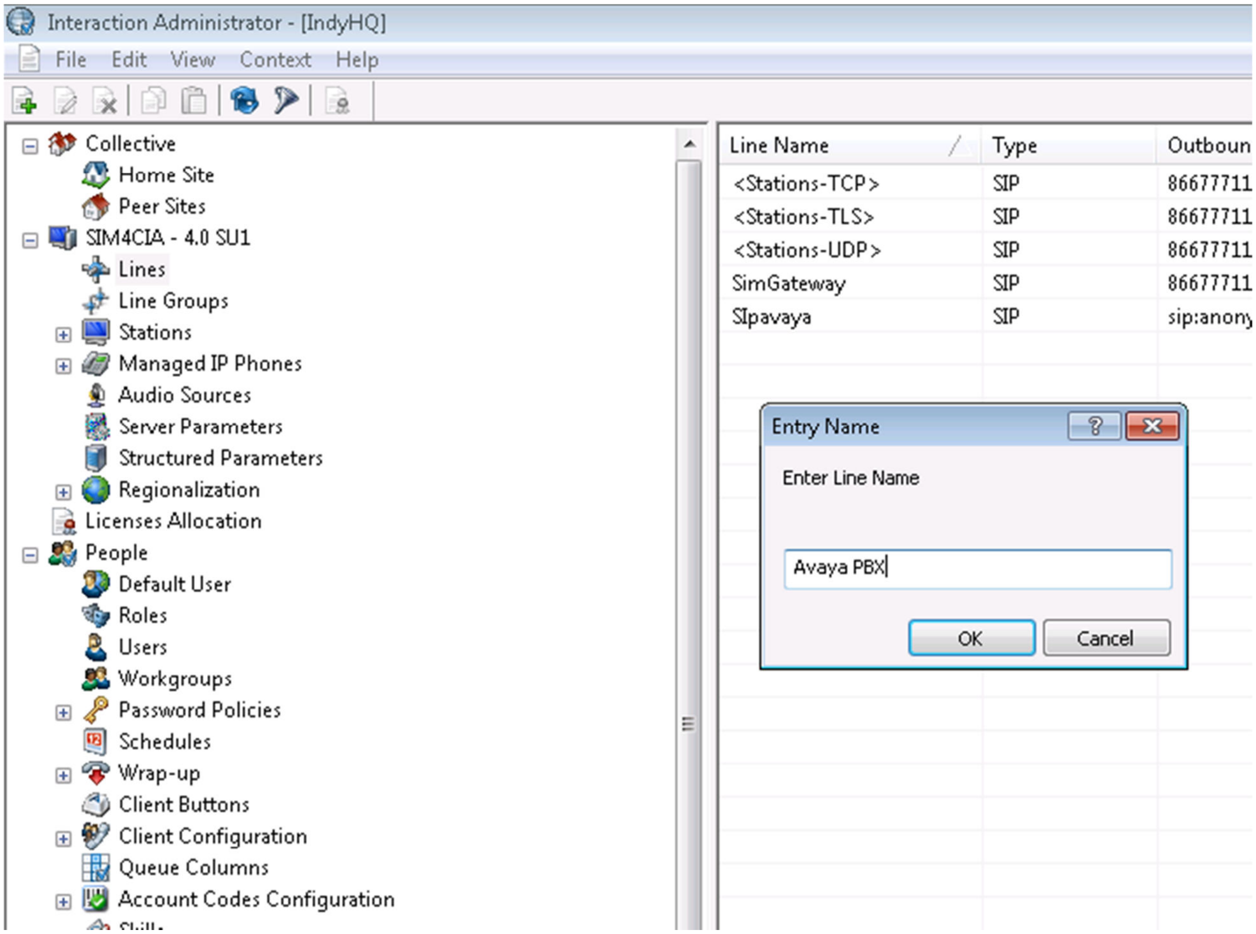
Avaya Condensed Configuration Notes

Communication Manager --> change node-names ip	Set Session Manager as a valid IP node
Communication Manager --> list ip-interfaces all	Note CM and SM SIP Interface IP addresses
Communication Manager --> change ip-codec-set 1	Set Codecs to G.711 MU and G.729
Communication Manager --> change ip-network-region 1	Change IP network region to authoritative domain
Communication Manager --> add signaling-group n	Create SIP Signaling groups for Session Manager
Communication Manager --> add trunk-group n	Create SIP Trunks for calls to Avaya and calls to CIC
Communication Manager --> change route pattern n	Configure Route Pattern for calls to Avaya and calls to CIC
Communication Manager --> change ars analysis n	Set Automatic Route Selection for calls from Avaya to CIC
Session Manager --> Domains	Set name to authoritative domain name and type to SIP

Session Manager --> Locations --> General:	Set descriptive name of CIC and any descriptive notes
Session Manager --> Locations --> Location Pattern:	Set IP Address Pattern the networks involved e.g. 10.64.1.*
Session Manager --> Routing --> Adaptations --> General Settings:	Module name drop down should be set to DigitConversionAdapter, Module Parameter as odstd=<address> where <address> is the IP of Session Manager
Session Manager --> Routing --> Adaptations --> Under Digit Conversion for Incoming Calls to SM:	Matching Patterns should be created to convert any dialed numbers from the CIC server i.e. 8001, and all conversion pattern entries
Session Manager --> Routing --> Adaptations --> Under Digit Conversion for Outgoing Calls from SM:	Set Matching Pattern for any dialed number from Avaya i.e. 3175555555 and all conversion pattern entries
Session Manager --> Routing --> SIP Entities:	CIC Server will need to be added as a SIP Entity
Session Manager --> Routing --> Entity Links:	Enter SIP entity names for CIC server and Session Manager, and input respective ports, trusts and protocols
Session Manager --> Routing --> Routing Policies:	Set routing policy of CIC server to time range of 24/7
Session Manager --> Routing --> Dial Patterns	Set dial patterns which will route to the CIC server and patterns which will route to the Communications Manager
System Manager --> Elements --> Session Manager --> Session Manager Administration	Set Session Manager name, description and IP, if not already done. Set Session Manager name, network mask, and default gateway in security settings

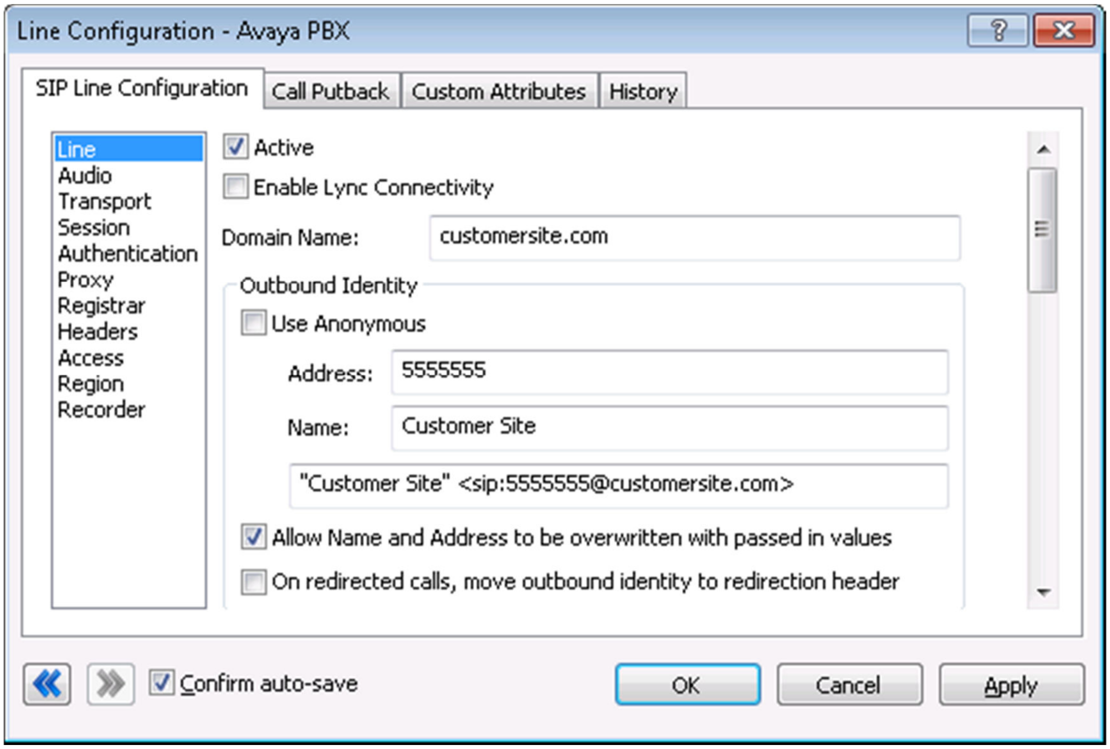
Chapter 2: xIC Setup

Step 1: Create the SIP Line



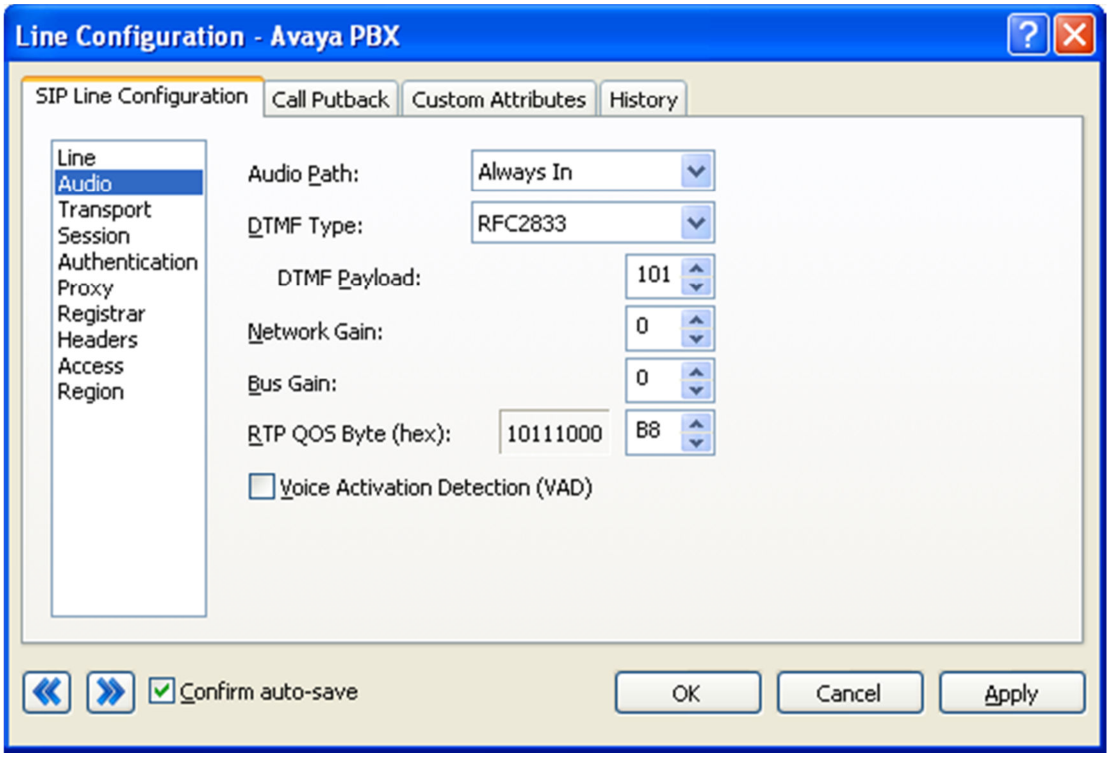
In Interaction Administrator, select lines, and then right click the right panel to create a new line.

Step 2: Create the SIP Line



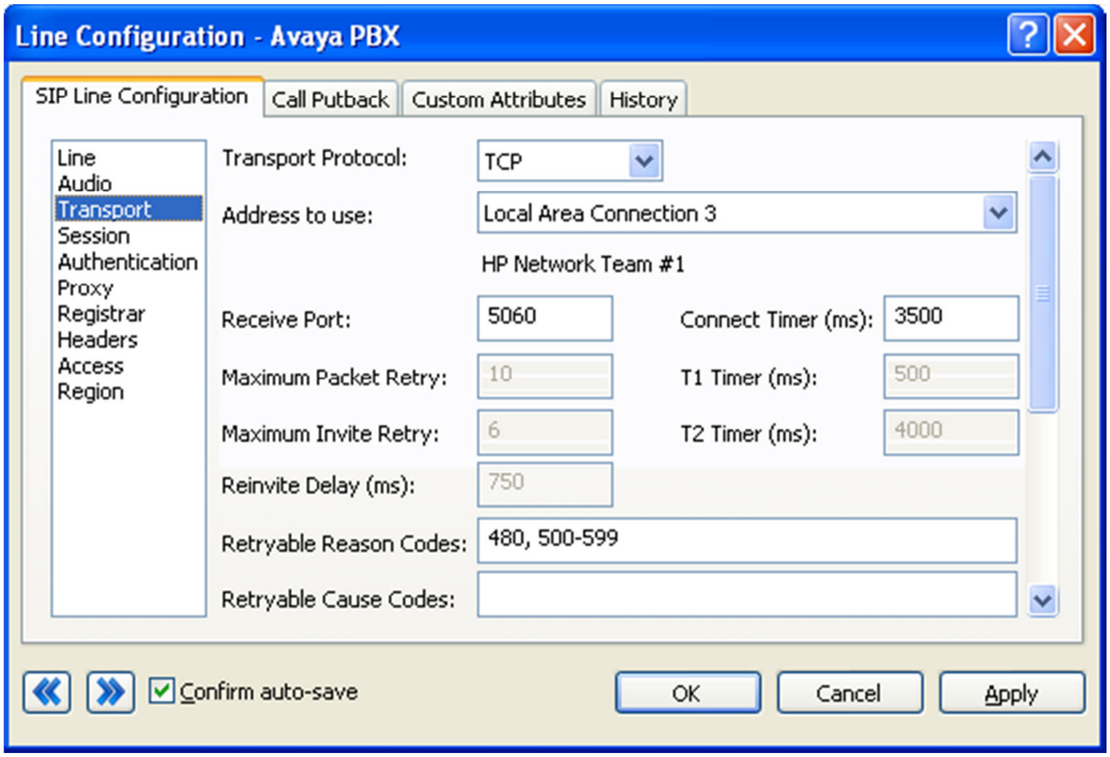
Take note of the Domain Name value here. It must match the value of the domain name in the Far-end Domain in the Avaya SIP Signaling Group. The domain name customersite.com is used here, although avaya.com is used in other examples.

Step 3: Configure Audio



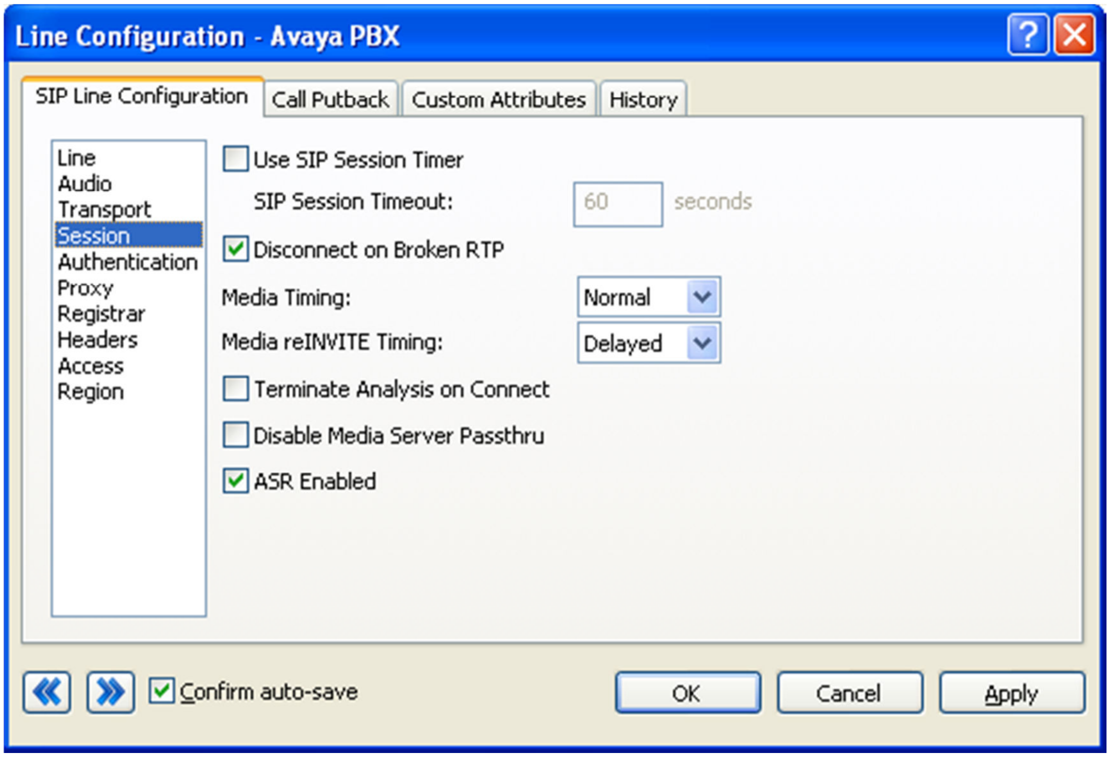
Use default settings. Audio Path should be set to Always In.

Step 4: Configure Transport Information



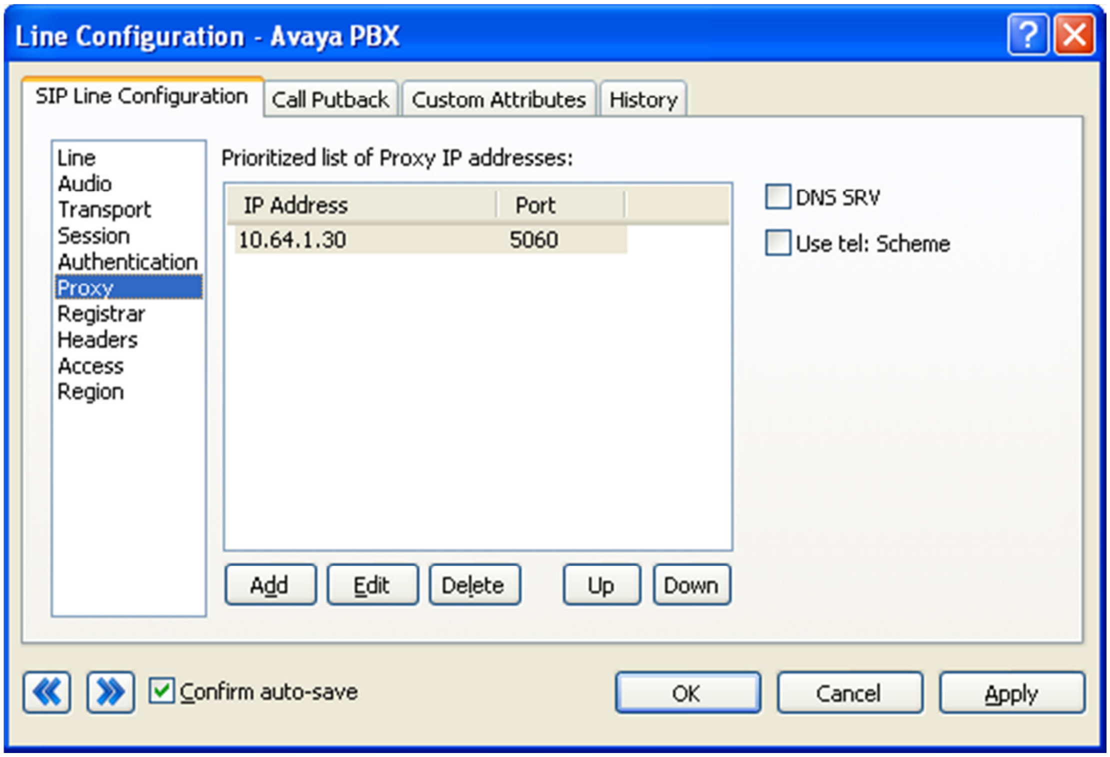
Use default settings, except use TCP as protocol.

Step 5: Configure Session Information



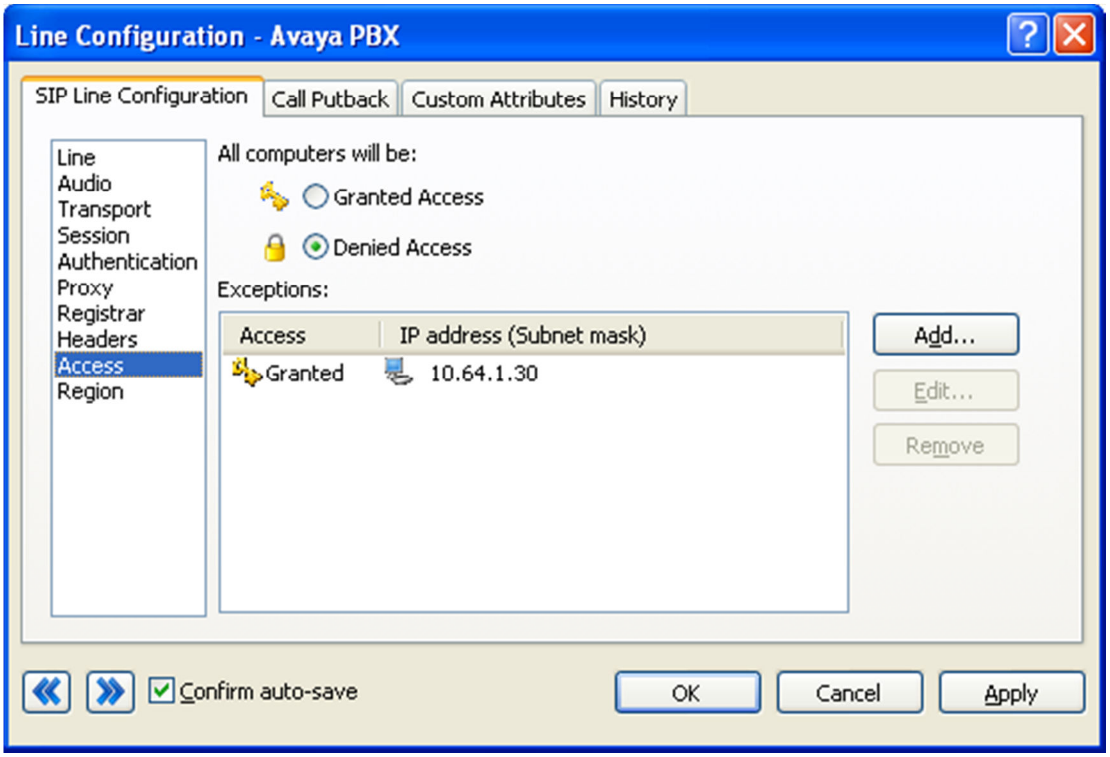
Use default settings.

Step 6: Configure Proxy Information



Set list of Proxy IP Addresses, which should match the IP of Avaya Session Manager.

Step 7: Configure Access List



Set Access List to “Denied Access” with the exception matching the IP of Avaya Session Manager.

Chapter 3: Communications Manager Setup

Step 1: Add CIC server as a valid IP Node

Configure the Session Manager as a valid IP node on the Communications Manager using a **node-name**. Use the **change node-names ip** command to add the **Name** and **IP Address** for the Session Manager. In the example, **SessionManager** and **10.64.1.30** were used.

change node-names ip		Page 1 of 2
	IP NODE NAMES	
Name	IP Address	
SessionManager	**10.64.1.30**	

Step 2: Note IP Interfaces:

Use the **list ip-interface all** command and note the **PROCR** interface address to be used for SIP trunks between the Communication Manager and the Session Manager

```
list ip-interface all
IP INTERFACES
Net
ON Type       Slot    Code/Sfx        Node Name/     Mask          Rgn    VLAN
                                      IP-Address/
                                      Gateway Node
-- ------ ----- --------              ---------------  ----          ---    ----
y PROCR                               procr          /25           1
                                      10.64.1.30

                                      ……
```

Step 3: Set Codecs

Use the **change ip-codec-set n** command to set codecs. Set Fax to T.38

```
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.729           n               2               20
3:
```

```
change ip-codec-set 1                                              Page 2 of 2
                              IP Codec Set
                                      Allow Direct-IP Multimedia? n

                        Mode                    Redundancy
        FAX             t.38-standard           0
        Modem           off                     0
        TDD/TTY         off                     0
```

Step 4: Configure IP Network Region

Use the **change ip-network-region n** command to configure IP network region.

```
change ip-network-region 1                                         Page 1 of 19 IP
                                NETWORK REGION
Region: 1
Location: 1                     Authoritative Domain: avaya.com
Name: Customer Site
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
```

Step 5: Add SIP Signaling Group

Use the **add signaling-group n** command, where n is an available signaling group number. Use the PROCR node name noted earlier for the near end group and the Session Manager name for the far end group.

```
add signaling-group 100                                          Page 1 of 1
                                SIGNALING GROUP

Group Number: 100              Group Type: sip
IMS Enabled? n            Transport Method: tcp
Q-SIP? n                                                     SIP Enabled LSP? n
IP Video? n                                           Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y      Peer Server: SM
Near-end Node Name: procr                 Far-end Node Name: SessionManager
Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                    Far-end Network Region: 1
Far-end Domain: avaya.com
                                              Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: allow                      RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload
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
```

Step 6: Configure SIP Trunk for calls within the Avaya Solution

Use the **add trunk-group n** command, where n is an available signaling group number. Use Signaling Group defined earlier, and define number of Members as SIP trunk count allocated to this trunk group.

```
add trunk-group 100                                                    Page 1 of 21
                              TRUNK GROUP
Group Number: 100             Group Type: sip                          CDR Reports: y
Group Name: Enterprise             COR: 1               TN: 1          TAC:
Direction: two-way            Outgoing Display? y
Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: public-ntwrk         Auth Code? n
                                        Member Assignment Method: auto
                                        Signaling Group: 100
                                        Number of Members: 100
```

```
add trunk-group 100                                         Page 3 of 21
                              TRUNK FEATURES
        ACA Assignment? n                 Measured: none
                                                            Maintenance Tests? y
                              Numbering Format: public
                                             UUI Treatment: service-provider

                                             Replace Restricted Numbers? n
                                             Replace Unavailable Numbers? n
```

Step 7: Configure SIP Trunk for calls to CIC

Add a second trunk group using the **add trunk-group n** command, where n is an available signaling group number. Use Signaling Group defined earlier, and define number of Members as SIP trunk count allocated to this trunk group.

```
add trunk-group 200                                                    Page 1 of 21
                                TRUNK GROUP
Group Number: 200               Group Type: sip                        CDR Reports: y
Group Name: CIC                      COR: 1               TN: 1        TAC:
Direction: two-way              Outgoing Display? y
Dial Access? n                                            Night Service:
Queue Length: 0
Service Type: public-network         Auth Code? n
                                          Member Assignment Method: auto
                                          Signaling Group: 100
                                          Number of Members: 100
```

```
add trunk-group 200                                               Page 3 of 21
                                TRUNK FEATURES
     ACA Assignment? n                    Measured: none
                                                            Maintenance Tests? y
                                Numbering Format: public
                                               UUI Treatment: service-provider

                                               Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
```

```
add trunk-group 200                                               Page 4 of 21
                                PROTOCOL VARIATIONS
                                Mark Users as Phone? n
                           Prepend '+' to Calling Number? n
                           Send Transferring Party Information? n
                                Network Call Redirection? n
                                Send Diversion Header? n
                                Support Request History? y
                           Telephone Event Payload Type: 101
```

Step 8: Configure Route Pattern for calls to Avaya

Use **change route pattern n** command, where **n** is an available route pattern.

```
change route-pattern 2                                                    Page 1 of 3
                        Pattern Number: 2        Pattern Name: toAvaya
                        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:   100  0                                                               n user
2:                                                                        n user
                                  .......
```

Step 9: Configure Route Pattern for calls to CIC

Use **change route pattern n** command, where **n** is an available route pattern.

change route-pattern 3									Page 1 of 3
				Pattern Number: 3			**Pattern Name: toCIC**		
				SCCAN? n			Secure SIP? n		
	Grp No	**FRL**	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del Dgts	Inserted Digits	DCS/ IXC QSIG Intw
1:	**200**	**0**							n user
2:									n user
						…….			

Step 10: Administer ARS Analysis

Here is a sample Automatic Route Selection (ARS). This can be used to direct call center calls dialed from the Avaya solution to the CIC server. In this example, the call center extension is 5001. More entries could be entered here depending on numbers needed to dial into call center.

```
change ars analysis 0                                                   Page 1 of 2 ARS
                              DIGIT ANALYSIS TABLE
                                        Location: all                 Percent Full: 1

      Dialed   Total          Route   Call     Node    ANI
      String   Min     Max    Pattern Type     Num     Reqd
      5001     4       4      3       pubu     n
                                      …….
```

Step 11: Save Communication Manager Settings

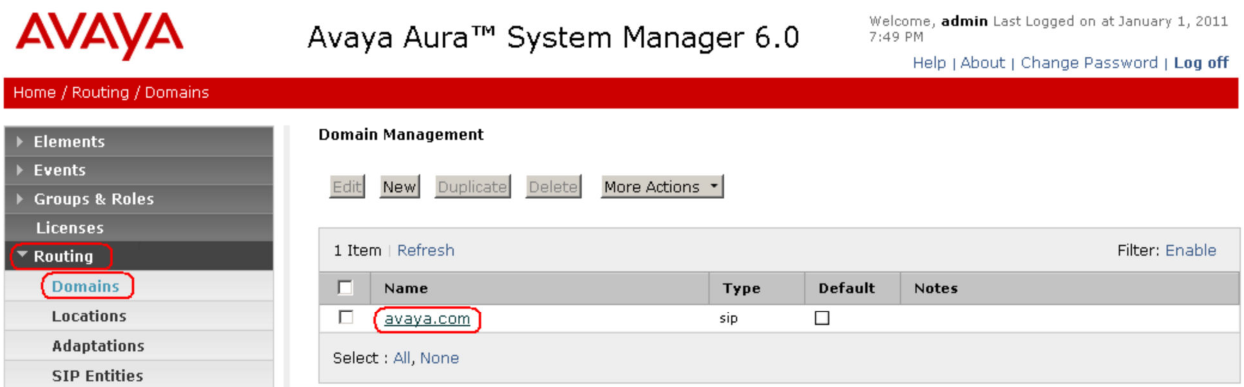
Use **save translation** command to save changes.

Chapter 4: Session Manager Setup

Step 1: Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right. The following screen will then be shown. Fill in the following fields and click **Commit**.

- **Name:** The authoritative domain name (e.g. **avaya.com**)
- **Type** Select **sip**
- **Notes:** Descriptive text (optional)



Step 2: Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. A single location is added to the configuration for Communication Manager and Interactive Intelligence CIC System Session Director.

To add a location, navigate the menu on the left **Routing** → **Locations** on the left and click on the **New** button on the right (not shown). The following screen will then be shown. Fill in the following:

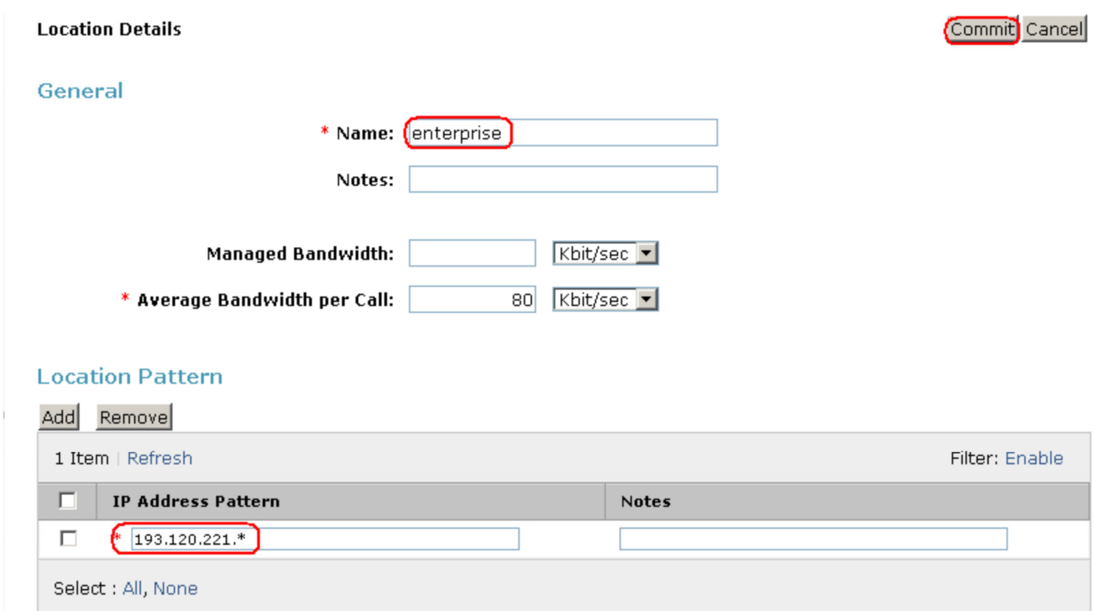
Under **General**:

- **Name:** A descriptive name
- **Notes:** Descriptive text (optional)
- **Managed Bandwidth:** Leave the default

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location. In these Application Notes, the pattern selected defined the networks involved e.g. **10.64.1.*** for referring the Enterprise network. **Note:** Other patterns can be used
- **Notes:** Descriptive text (optional)

The screen below shows addition of the **Enterprise** location, which includes all the components of the compliance environment. Click **Commit** to save.

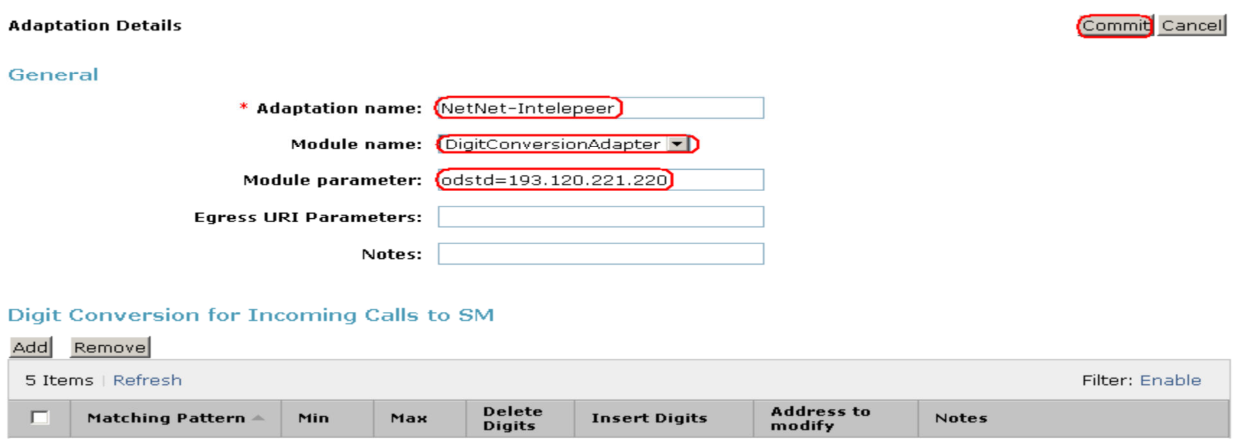


Step 3: Add Adaptations

In order to maintain digit manipulation centrally on Session Manager, an adaptation module has to be configured with numbering plan offered from the Service Provider. To add an adaptation, under the **Routing** → **Adaptations** on the left and click on the **New** button on the right (not shown). The following screen will then be shown. Fill in the following:

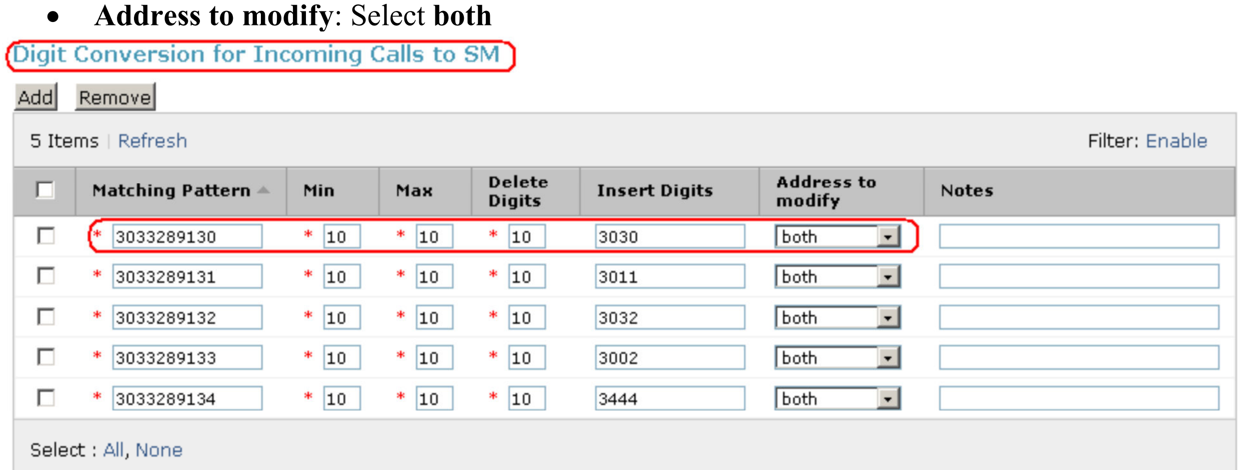
Under **General**:

- **Name:** A descriptive name i.e: **CIC**
- **Module Name:** From the dropdown list select **DigitConversionAdapter**
- **Module Parameter:** Enter **odstd=<address>** where address is the IP address of the SIP interface of Session Manager



Under **Digit Conversion for Incoming Calls to SM**:

- **Matching Pattern**: The dialed number from the CIC server i.e. **8001**
- **Min/Max**: Minimum/Maximum number of digits i.e. **4**
- **Delete**: Digits to be deleted i.e. **4**
- **Insert Digits**: Digit to be added i.e. **8001**
- **Address to modify**: Select **both**



Under **Digit Conversion for Outgoing Calls from SM**:

- **Matching Pattern**: The dialed number from Avaya network i.e. **3175555555**
- **Min:/ Max**: Minimum/ Maximum number of digits i.e. **10**
- **Delete**: Digits to be deleted i.e. **10**
- **Insert Digits**: Digit to be added i.e. **3175555555**
- **Address to modify**: Select **both**

Note: This Digit Conversion rule was used by Session Manager to modify outgoing SIP messages to match the format expected by the CIC server. Standard CIC Dial Plan rules can be configured to match patterns delivered from Avaya Session Manager

Digit Conversion for Outgoing Calls from SM

Add Remove

5 Items | Refresh Filter: Enable

☐	Matching Pattern	Min	Max	Delete Digits	Insert Digits	Address to modify	Notes
☐	* 3033289130	* 10	* 10	* 10	3033289130	both	
☐	* 3033289131	* 10	* 10	* 10	3033289131	both	
☐	* 3033289132	* 10	* 10	* 10	3033289132	both	
☐	* 3033289133	* 10	* 10	* 10	3033289133	both	
☐	* 3033289134	* 10	* 10	* 10	3033289134	both	

Select : All, None

Step 4: Add SIP Entities

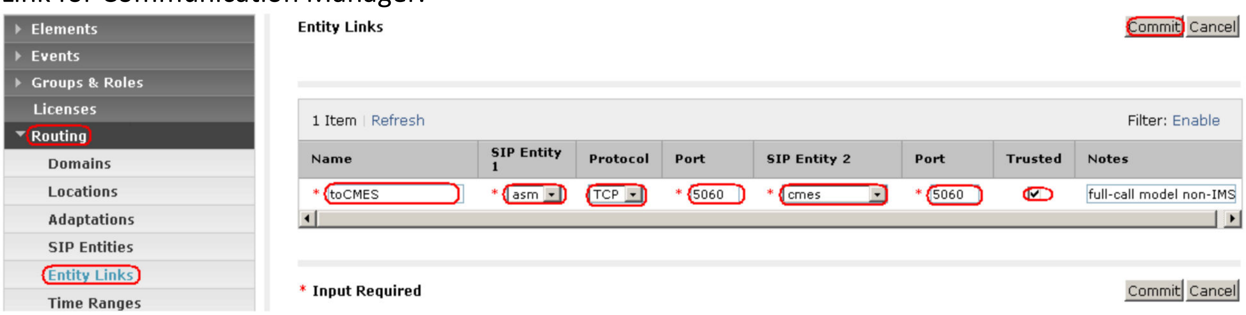
A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In this scenario the “SIP Entity” will be the CIC system. In the sample configuration, a SIP Entity is added for the Session Manager, the PROC interface on the Communication Manager and the SIP Trunking for Interactive Intelligence CIC System (CIC) which acts as gateway with the Service Provider.

Step 5: Add Entity Links

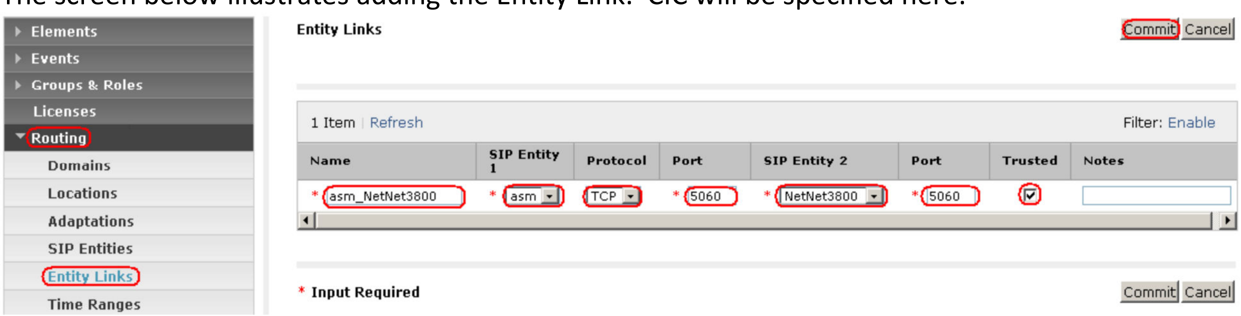
A SIP trunk between Session Manager and a telephony system is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name
- **SIP Entity 1:** Select the **SessionManager** entity
- **Port:** Port number to which the other system sends SIP requests
- **SIP Entity 2:** Select the **CIC** entity
- **Port:** Port number on which the other system receives SIP requests
- **Trusted:** Check this box, otherwise calls from the associated SIP Entity specified will be denied
- **Protocol:** Select the transport protocol between **UDP/TCP/TLS** to align with the definition on the CIC server. In these Application Notes **TCP** was used.

Click **Commit** to save each Entity Link definition**.** The following screen illustrates adding the Entity Link for Communication Manager.



The screen below illustrates adding the Entity Link. CIC will be specified here.

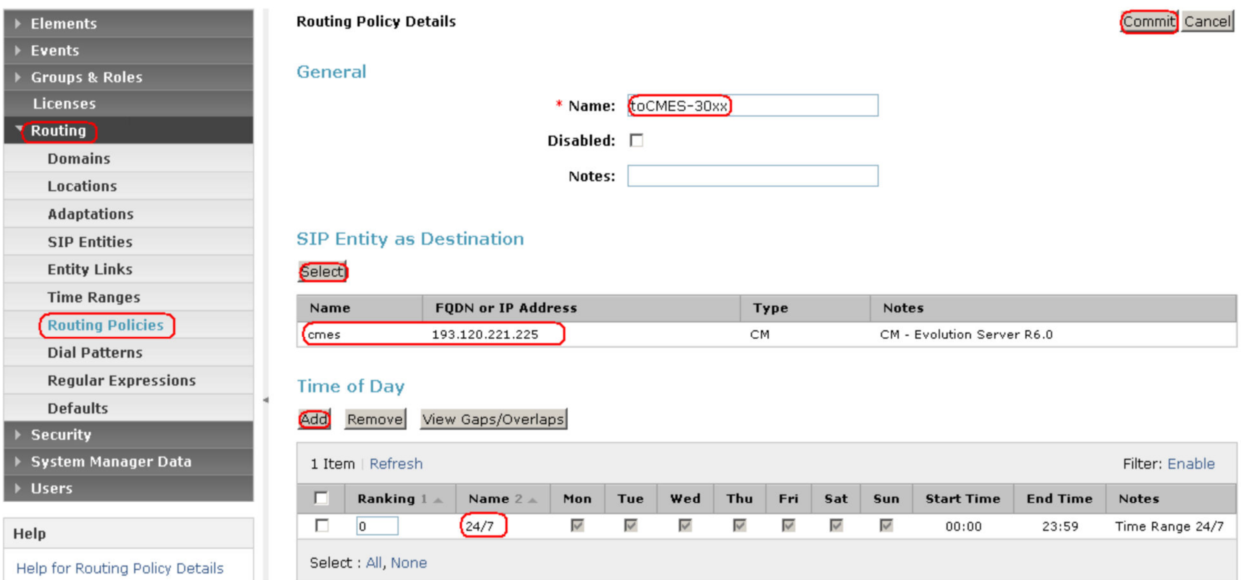


Step 6: Add Routing Policies

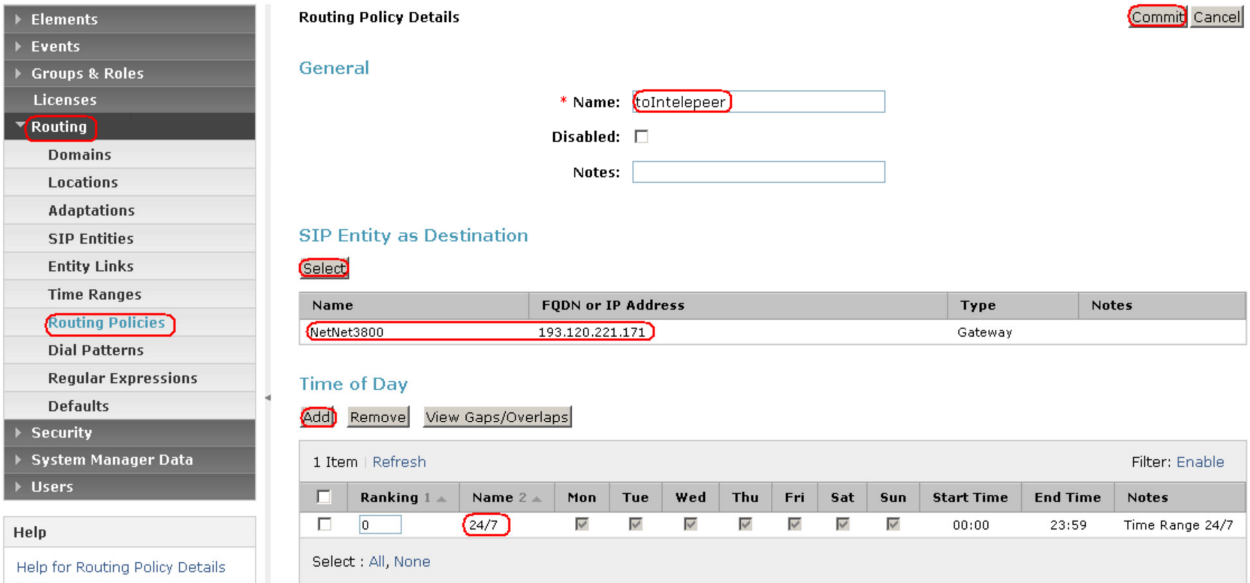
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Add Adaptations**. Two routing policies must be added: one for Communication Manager Evolution Server and one for the CIC Server. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

- Under General
  - Enter a descriptive name in **Name**
- Under **SIP Entity as Destination**
  - Click **Select**, and then select the appropriate SIP entity to which this routing policy applies
- Under **Time of Day:**
  - Click **Add**, and select the time range configured. In these Application Notes, the predefined **24/7** Time Range is used.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy Input. The CIC server should be defined as a SIP Entity with Name of CIC and FQDN or IP address.



Elements
Events
Groups & Roles
Licenses
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Security
System Manager Data
Users
Help
Help for Routing Policy Details
Routing Policy Details
Commit
Cancel
General
* Name: toIntelepeer
Disabled:
Notes:
SIP Entity as Destination
Select
Name
FQDN or IP Address
Type
Notes
NetNet3800
193.120.221.171
Gateway
Time of Day
Add
Remove
View Gaps/Overlaps
1 Item | Refresh
Filter: Enable
Ranking 1
Name 2
Mon
Tue
Wed
Thu
Fri
Sat
Sun
Start Time
End Time
Notes
0
24/7
00:00
23:59
Time Range 24/7
Select : All, None

Step 7: Add Dial Patterns

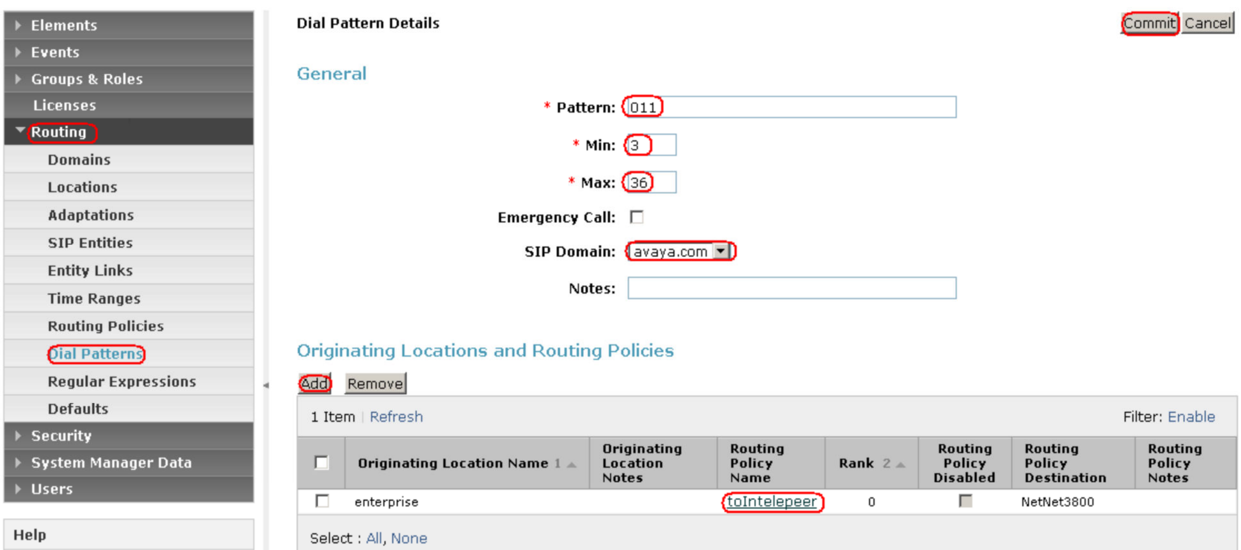
Dial patterns must be defined that will direct calls to the CIC server or the Communications Manager. For example, 4-digit extensions beginning with **8** may reside on Communication Manager and the number **3175555555** may be one number which routes to the CIC. Dial patterns for the CIC Server should be added here.

To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right (not shown). Fill in the following, as shown in the screen below, with entries corresponding to dial patterns which should route to the CIC Server:

Under **General**:

- **Pattern:** Dialed number or prefix i.e. **3175555555**
- **Min:** Minimum length of dialed number i.e. **10**
- **Max:** Maximum length of dialed number i.e. **10**
- **SIP Domain:** Select **avaya.com**

Under **Originating Locations and Routing Policies**, click **Add**, and then select the appropriate location and routing policy from the list. Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows a sample the dial pattern definition for SIP Trunking service. Create as many dial pattern required for the destination considered.



Repeat the process adding one or more dial patterns for routes and extensions that reside on both the CIC server and Communication Manager. Fill in the following, which corresponds to the dial pattern for routing calls to Communication Manager:

Under **General**:

- **Pattern:** Dialed number or prefix i.e. **8**

- **Min:** Minimum length of dialed number i.e. **4**
- **Max:** Maximum length of dialed number i.e. **4**
- **SIP Domain:** Select **avaya.com**

Under **Originating Locations and Routing Policies**, click **Add**, and then select the appropriate location and routing policy from the list. Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows a sample the dial pattern definition for Communication Manager Evolution Server. The figure below summarizes the creation of several dial patterns created for the compliance test bed.

Dial Patterns

Edit | New | Duplicate | Delete | More Actions | Commit

10 Items | Refresh   Filter: Enable

☐	Pattern	Min	Max	Emergency Call	SIP Domain	Notes
☐	00	2	36	☐	avaya.com	
☐	011	3	36	☐	avaya.com	
☐	0766878xxx	10	10	☐	avaya.com	i/c to 30xx
☐	1	11	11	☐	avaya.com	
☐	303	3	36	☐	avaya.com	
☐	30xx	4	4	☐	avaya.com	
☐	3111	4	4	☐	avaya.com	
☐	3444	4	4	☐	avaya.com	toVP
☐	7	2	36	☐	avaya.com	
☐	911	3	3	☑	avaya.com	

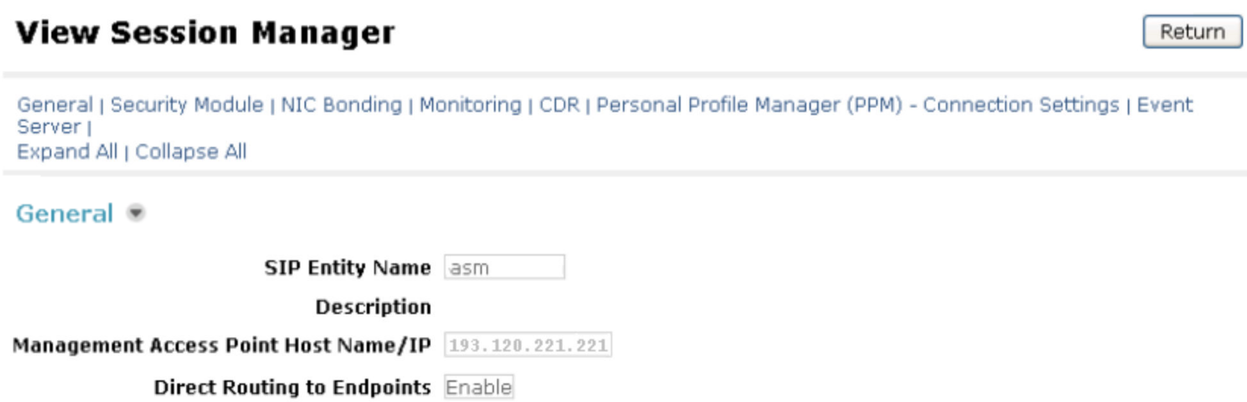
Select : All, None

Step 8: Add/View Avaya Aura Session Manager

The creation of a Session Manager element provides the linkage between System Manager and Session Manager. This was most likely done as part of the initial Session Manager installation. To add a Session Manager, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane (**Section 6.1**) and click on the New button in the right pane (not shown). If the Session Manager already exists, click View (not shown) to view the configuration. Enter/verify the data as described below and shown in the following screen: In the General:

Under **General**:

- **SIP Entity Name**: Select the name of the SIP Entity added for Session Manager
- **Description**: Descriptive comment (optional)
- **Management Access Point Host Name/IP**: Enter the IP address of the Session Manager management interface



In the **Security Module** section, (not shown) enter the following values:

- SIP Entity IP Address: Should be filled in automatically based on the SIP Entity Name. Otherwise, enter IP address of Session Manager signaling interface.
- Network Mask: Enter the network mask corresponding to the IP address of Session Manager.
- Default Gateway: Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields. Click **Save** to add this Session Manager.

Chapter 5: Customer Interaction Center 4.0 Validation Test Matrix

Testing the Core Feature Set

The following table contains a set of tests for assessing the functionality of the UM core feature set. The results are recorded as either:

- Pass (**P**)
- Conditional Pass (**CP**)
- Fail (**F**)
- Not Tested (**NT**)
- Not Applicable (**NA**)

Refer to:

- Appendix for a more detailed description of how to perform each call scenario.

No.	Call Scenarios (See Appendix for Detail)	(P/CP/F/NT)	Reason for Failure
1	Dial Auto Attendant (AA). Dial the extension for the AA and confirm the AA answers the call.		
2	Send a test Fax message to user extension. Confirm the Fax is received in the user's inbox.		
3	Call Transfer by Dial By Name.		
4	Call Transfer by Dial By Name and have the called party answer. Confirm the correct called party answers the phone.		
5	Call Transfer by Dial By Name when the called party's phone is busy.		
6	Call Transfer by Dial by Name when the called party does not answer.		
7	Blind Transfer		
8	Consult Transfer		
9	Call Audio Recorded		

Appendix

Dial Auto Attendant (AA)

1. Create an Auto Attendant using the CIC Web Administrator:
2. Dial the extension of Auto Attendant.
3. Confirm the AA answers the call.

Call Transfer by Dial By Name

1. Dial the pilot number for the CIC server from a phone that is NOT associated with a CIC user.
2. To search for a user by name:
    a. Press 2 to Dial By Name.
    b. Call Transfer by Dial By Name by entering the name of an CIC user using the telephone keypad, last name first.

Note: Even though some keys are associated with three or four numbers, for each letter, each key only needs to be pressed once regardless of the letter you want.
Ignore spaces and symbols when spelling the name.Called Party Answers

Call Transfer by Dial By Name to a user in the same dial plan and have the called party answer.

1. Confirm the call is transferred successfully.

Called Party is Busy

1. Call Transfer by Dial By Name to a user in the same dial plan when the called party is
    1. busy.
2. Confirm the calling user is routed to the correct voicemail.

Called Party does not Answer

1. Call Transfer by Dial By Name to a user in the same dial plan and have the called party not
    2. answer the call.
2. Confirm the calling user is routed to the correct voicemail.

Testing Fax Features

To test fax functionality:

1. Dial the extension for a fax-enabled CIC user from a fax machine.
2. Confirm the fax message is received in the user's inbox.

Note: You may notice that the CIC server answers the call as though it is a voice call (i.e. you will hear: "Please leave a message for…"). When the CIC server detects the fax CNG tones, it switches into fax receiving mode, and the voice prompts terminate.
Note: CIC only supports T.38 for sending fax.

Blind Transfer

1. Verify Putback is enabled on SIP Line.
2. Ring No Answer to a user's voicemail.
3. Zero Out to user's Operator.
4. Answer Operator phone.
5. Verify that all resources are released from IC.

Consult Transfer

1. Verify Putback is enabled on SIP Line, and that Follow Me is enabled for your test user.
2. Ring No Answer to a user's voicemail.
3. Press 2 to Follow Me.

Note: Follow me should be setup to an internal extension.

4. Answer Follow Me call.
5. Verify that all resources are released from IC.

Audio Recording

1. Perform ad hoc recording from the client using an Avaya as the station
2. Ensure calls have been recorded and delivered to user's email.