Avaya Solution & Interoperability Test Lab

Application Notes for IPC System Interconnect 15.03 with Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager 6.1 using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for IPC System Interconnect 15.03 to interoperate with Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager 6.1 using SIP trunks.

IPC System Interconnect is a trading communication solution. In the compliance testing, IPC System Interconnect used SIP trunks to Avaya Aura® Session Manager, for turret users on IPC to reach users on Avaya Aura® Communication Manager and on the PSTN.

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

These Application Notes describe the configuration steps required for IPC System Interconnect 15.03 to interoperate with Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager 6.1 using SIP trunks.

IPC System Interconnect is a trading communication solution. In the compliance testing, IPC System Interconnect used SIP trunks to Avaya Aura® Session Manager, for turret users on IPC to reach users on Avaya Aura® Communication Manager and on the PSTN.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established among IPC turret users with Avaya SIP, Avaya H.323, and/or PSTN users. Call controls were performed from the various users to verify the call scenarios.

The serviceability test cases were performed manually by disconnecting and reconnecting the LAN connection to the IPC ESS server.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included basic call, display, G.711,G.729, codec negotiation, media shuffling, hold/reconnect, DTMF, call forwarding unconditional/ring-no-answer/busy, blind/attended transfer, and attended conference.

The serviceability testing focused on verifying the ability of IPC System Interconnect to recover from adverse conditions, such as disconnecting/reconnecting the LAN connection to IPC System Interconnect.

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.

2.2. Test Results

All test cases were executed and verified.

2.3. Support

Technical support on IPC System Interconnect can be obtained through the following:

- **Phone:** (800) NEEDIPC, (203) 339-7800
- **Email:** systems.support@ipc.com

3. Reference Configuration

As shown in the test configuration below, IPC System Interconnect at the Remote Site consists of the Enterprise SIP Server (ESS), Alliance MX, System Center, and Turrets. SIP trunks are used from System Interconnect to Avaya Aura® Session Manager, to reach users on Avaya Aura® Communication Manager and on the PSTN. In the compliance testing, the "avaya.com" domain was used for Avaya site, and "ipc.com" was used on IPC site.

A five digit Uniform Dial Plan (UDP) was used to facilitate dialing between the Central and Remote sites. Unique extension ranges were associated with Avaya Aura® Communication Manager users at the Central site (20xxx), and IPC turret users at the Remote site (332xx).

The configuration of Avaya Aura® Session Manager is performed via the web interface of Avaya Aura® System Manager. The detailed administration of basic connectivity between Avaya Aura® Communication Manager, Avaya Aura® System Manager, and Avaya Aura® Session Manager is not the focus of these Application Notes and will not be described.

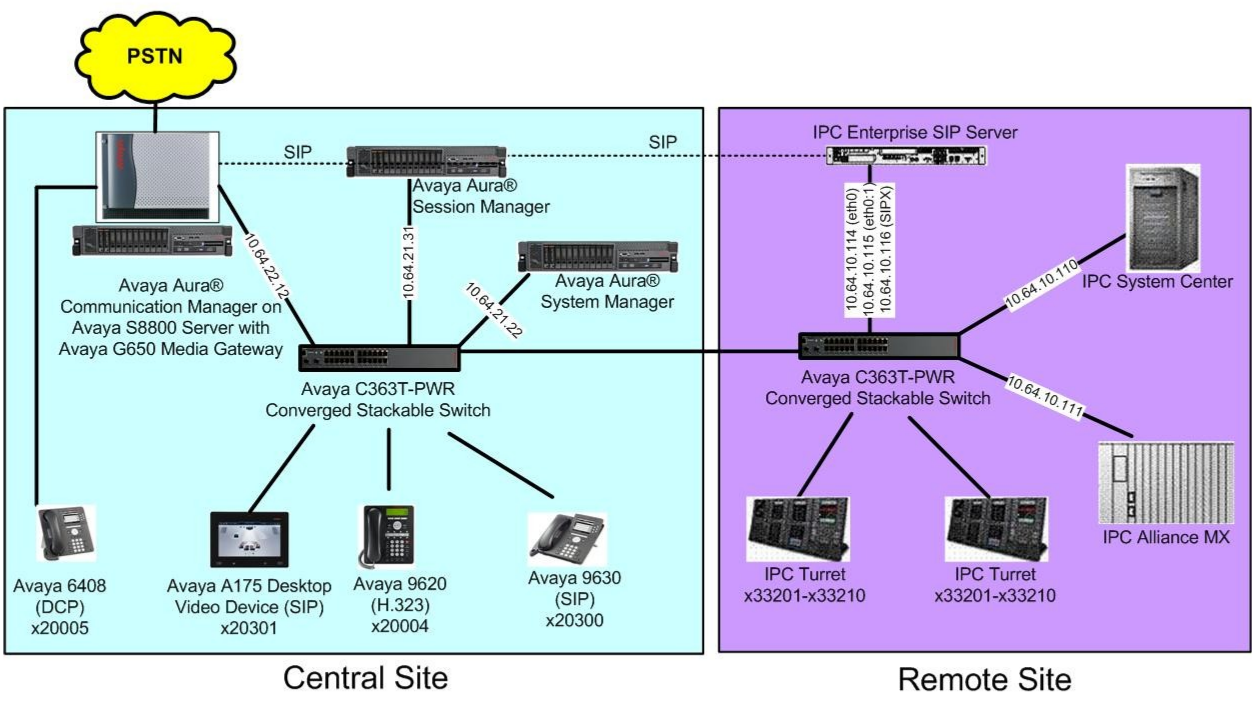


Figure 1: Test Configuration of IPC Alliance

4. Equipment and Software Validated

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

Equipment	Software
Avaya Aura® Communication Manager on Avaya S8800 Server	6.0.1(R016x.00.1.510.1) with special patch 19823
Avaya G650 Media Gateway • TN799DP C-LAN Circuit Pack • TN2302AP IP Media Processor • TN464F	 HW01 FW038 HW20 FW122 000010
Avaya Aura® Session Manager	6.1.5
Avaya Aura® System Manager	6.1.5
Avaya 9620 IP Telephone (H.323)	3.1
Avaya 9630 IP Telephone (SIP)	2.6.4
Avaya A175 Desktop Video Device (SIP)	1.0.2
IPC System Interconnect • Alliance MX • System Center o SIPX Line Card • Turrets • Enterprise SIP Server	15.03.00.07a 15.03.00.07a 15.03.00.07a 15.03.00.07a 15.03.00.07a 2.01.00-01

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify Communication Manager license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer uniform dial plan
- Administer AAR analysis
- Administer ISDN trunk group
- Administer tandem calling party number

In the compliance testing, the same set of codec set, network region, trunk group, and signaling group were used for the Avaya SIP and IPC turret users, which enabled IPC turret users to use the same digits dialing as Avaya SIP users, to reach other users on Communication Manager and on the PSTN.

5.1. Verify Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options                        Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 98
           Maximum Concurrently Registered IP Stations: 18000 1
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 1
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 376
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Administer System Parameters Features

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers.

This feature is needed to be able to transfer an incoming call from IPC back out to IPC (incoming trunk to outgoing trunk), and to transfer an outgoing call to IPC to another outgoing call to IPC (outgoing trunk to outgoing trunk). For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class Of Restriction or Class Of Service levels. Refer to [1] for more details.

```
change system-parameters features                              Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y

              Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n

             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

5.3. Administer SIP Trunk Group

Use the "change trunk-group n" command, where "n" is the existing SIP trunk group number used to reach Session Manager, in this case "8".

For **Group Name**, update as desired to reflect the same trunk group used to reach Session Manager and IPC. For **Number of Members**, enter sufficient number for simultaneous calls to Avaya SIP and IPC users. Note that a call between an Avaya SIP user and an IPC user uses two SIP trunks, whereas a call between an Avaya non-SIP user and an IPC user uses one SIP trunk. Make a note of the **Signaling Group** number.

```
change trunk-group 8                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 8                    Group Type: sip          CDR Reports: y
  Group Name: PN1 to SM_21_31             COR: 1        TN: 1        TAC: *008
   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: 8
                                                     Number of Members: 10
```

Navigate to **Page 3**, and enter "private" for **Numbering Format**.

```
change trunk-group 8                                              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
```

Navigate to **Page 4**, and enter "101" for **Telephone Event Payload Type**, as required by IPC.

```
change trunk-group 8                                              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
```

5.4. Administer SIP Signaling Group

Use the "change signaling-group n" command, where "n" is the existing SIP signaling group number used by the SIP trunk group from **Section 5.3**.

For **DTMF over IP**, enter "rtp-payload". For **Direct IP-IP Audio Connections**, enter "y". Make a note of the **Far-end Network Region** number, and the **Far-end Domain** value. Note that **Transport Method** is set to "tcp" for troubleshooting purpose, also note the values of **Near-end Listen Port** and **Far-end Listen Port**, which will be used later.

```
change signaling-group 8                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 8              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: CLAN1A                   Far-end Node Name: SM_21_31
 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: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 120                    IP Audio Hairpinning? y
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

5.5. Administer IP Network Region

Use the "change ip-network-region n" command, where "n" is the existing far-end network region number used by the SIP signaling group from **Section 5.4**.

For **Name**, update as desired to reflect the same network region used to reach IPC. Enter "yes" for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. In the compliance testing, the same network region was used for all Avaya users. Make a note of the **Codec Set** number.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name: PN1
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
```

5.6. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the existing codec set number used by the IP network region from **Section 5.5**. Update the audio codec types in the **Audio Codec** fields as necessary. For **Media Encryption**, make certain "none" is specified (not shown).

In the compliance testing, the same codec set was used for all Avaya users.

```
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:
 3:
 4:
 5:
 6:
 7:
```

5.7. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is the existing route pattern number to reach Session Manager, in this case "8". For **Pattern Name**, update as desired to reflect the same route pattern used to reach Session Manager and IPC.

```
change route-pattern 8                                          Page   1 of   3
                    Pattern Number: 8    Pattern Name: toSM61
                             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: 8    0                                                               n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                           none
```

5.8. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to IPC. Add an entry for the trunk group defined in **Section 5.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 2 and routed to trunk group 8 will result in a 5-digit calling number. The calling party number will be in the SIP "From" header.

```
change private-numbering 0                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext                Trk        Private           Total
Len Code               Grp(s)     Prefix            Len
 5  2                  8                            5      Total Administered: 4
 5  2                  12                           5         Maximum Entries: 540
```

5.9. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 332xx to IPC. Note that other methods of routing may be used. Use the "change uniform-dialplan 0" command, and add an entry to specify the use of AAR for routing digits 332xx, as shown below.

```
change uniform-dialplan 0                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                            Percent Full: 0

  Matching                    Insert              Node
  Pattern        Len  Del     Digits      Net Conv Num
 332               5    0                 aar   n
 333               5    0                 aar   n
```

5.10. Administer AAR Analysis

Use the “change aar analysis 0” command, and add an entry to route calls to 332xx. In the example shown below, calls with digits 332xx will be routed using route pattern “8”. Set the **Call Type** to “unku”, to prevent “+” being added as a prefix.

```
change aar analysis 3                                               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
    332                      5     5      8        unku           n
    333                      5     5      9        aar            n
```

5.11. Administer ISDN Trunk Group

Use the “change trunk-group n” command, where “n” is the existing ISDN trunk group number used to reach the PSTN, in this case “99”. Navigate to **Page 3**.

For **Modify Tandem Calling Number**, enter “tandem-cpn-form” to allow for the calling party number from IPC to be modified. By enabling this feature, the calling party number will be sent to PSTN when call is coming from IPC side via a SIP trunk.

```
change trunk-group 99                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                                         Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n                            Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: n
Network Call Redirection: none                    Hold/Unhold Notifications? n
             Send UUI IE? y     Modify Tandem Calling Number: tandem-cpn-form
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                         Ds1 Echo Cancellation? n

    Apply Local Ringback? n               US NI Delayed Calling Name Update? n
 Show ANSWERED BY on Display? y
                             Network (Japan) Needs Connect Before Disconnect? n
```

5.12. Administer Tandem Calling Party Number

Use the "change tandem-calling-party-num" command, to define the calling party number to send to the PSTN for tandem calls from IPC turret users.

In the example shown below, all calls originating from a 5-digit extension beginning with 3 and routed to trunk group 99 will result in a 10-digit calling number. For **Number Format**, use an applicable format, in this case "pub-unk".

```
change tandem-calling-party-num                                 Page   1 of   8
                  CALLING PARTY NUMBER CONVERSION
                          FOR TANDEM CALLS
     CPN                Trk                                Number
 Len Prefix             Grp(s)      Delete  Insert         Format

 5   3                  99          all     3035381202       pub-unk
```

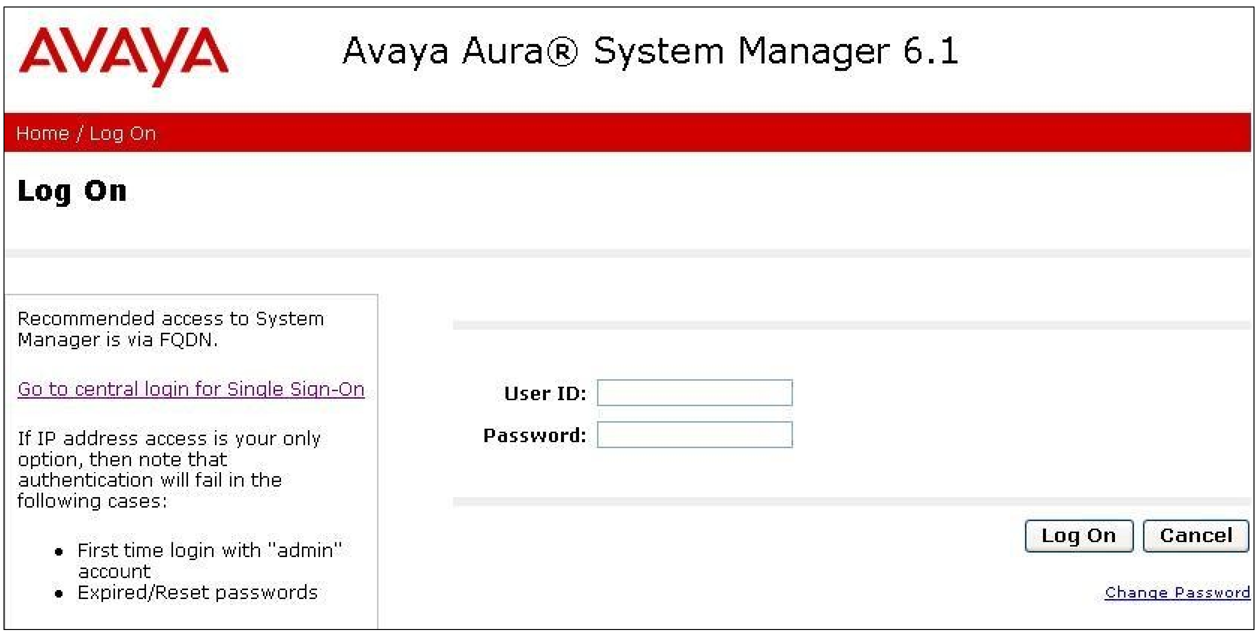
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Avaya Aura® Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer locations
- Administer adaptations
- Administer SIP entities
- Administer entity links
- Administer routing policies
- Administer dial patterns

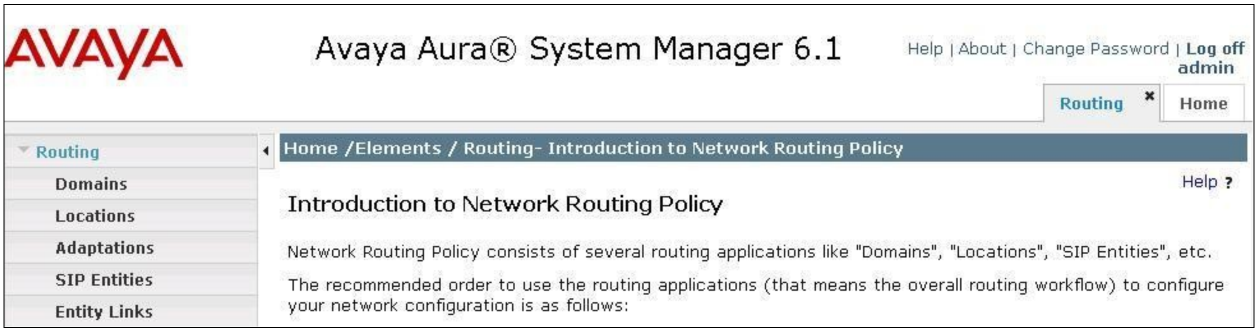
6.1. Launch System Manager

Access the System Manager web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the System Manager server. Log in using the appropriate credentials.

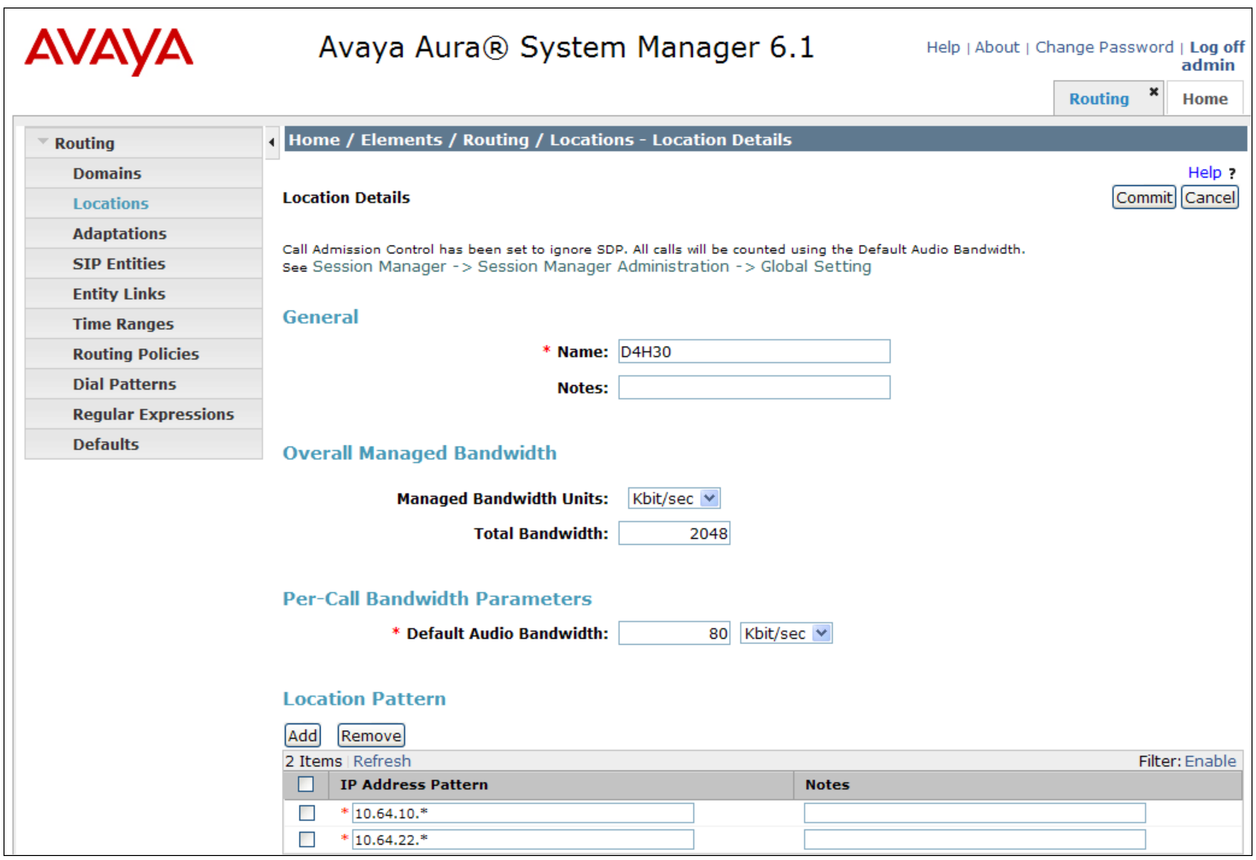


6.2. Administer Locations

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for IPC.



The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. In the **Location Pattern** sub-section, click **Add** and enter the applicable **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

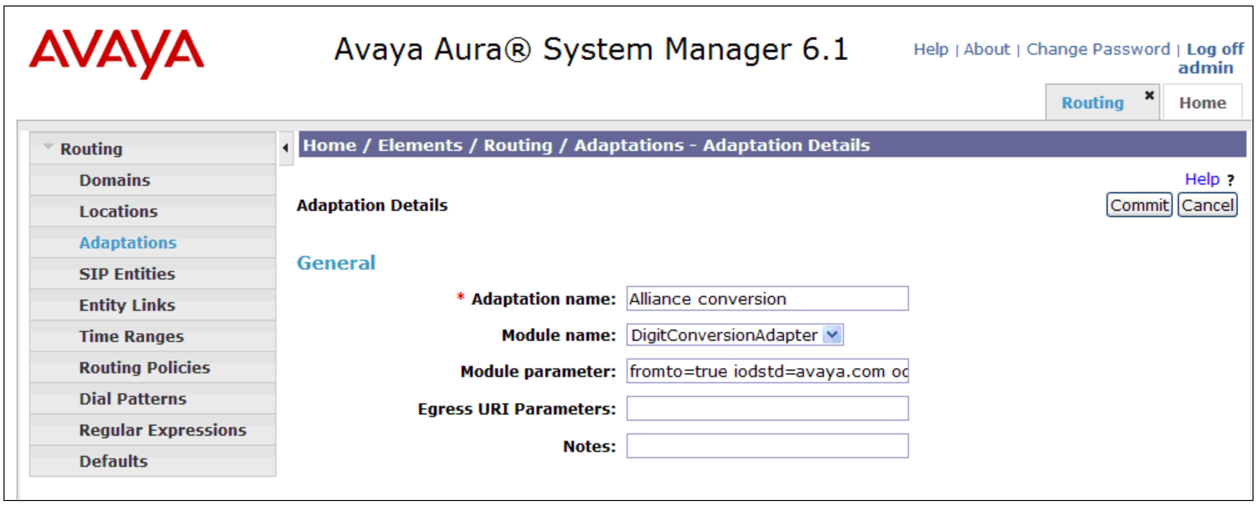


6.3. Administer Adaptations

Select **Routing → Adaptations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new adaptation for IPC.

The **Adaptation Details** screen is displayed. In the **General** sub-section, enter a descriptive **Adaptation name**. For **Module name**, select "DigitConversionAdapter".

For **Module parameter**, enter "iodstd=avaya.com odstd=ipc.com", where "avaya.com" is the Avaya side domain, and "ipc.com" is IPC side domain. This will set the source and destination domains for all incoming and outgoing calls for IPC.

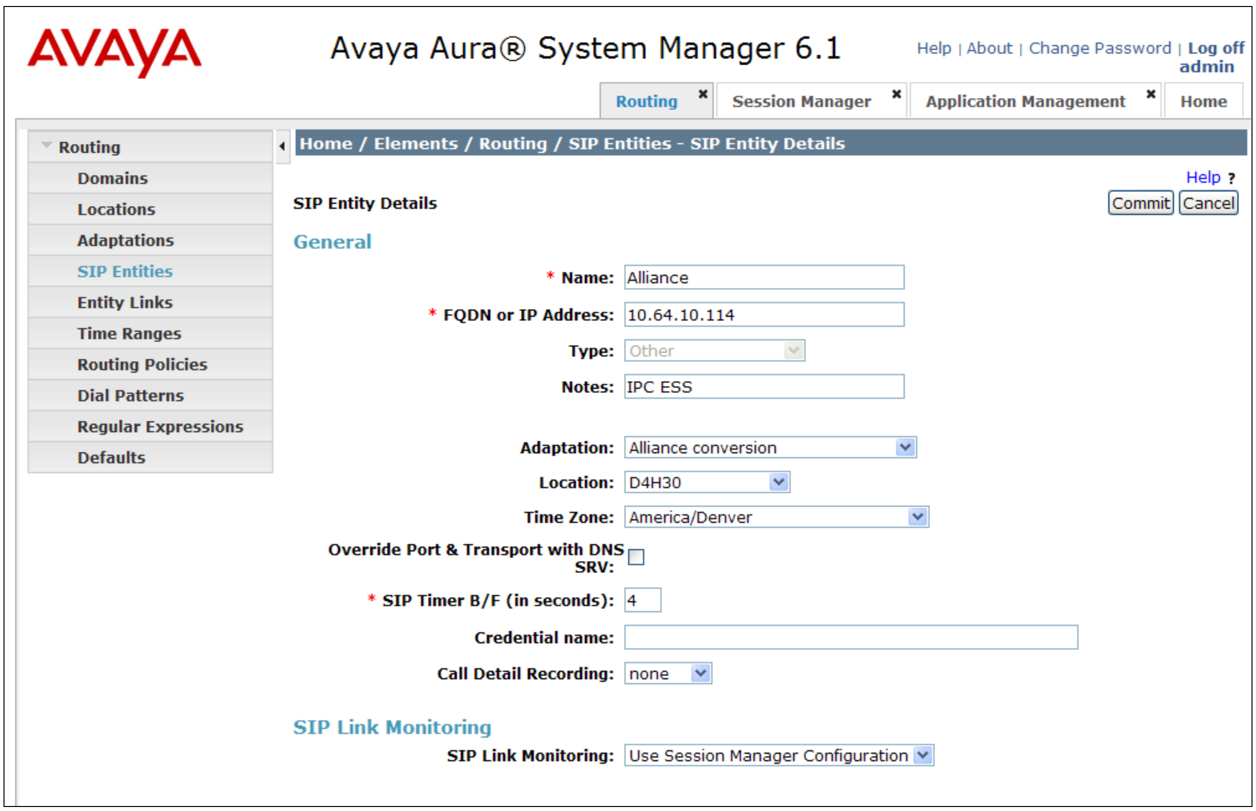


6.4. Administer SIP Entities

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for IPC.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the IPC ESS server.
- **Type:** "Other"
- **Adaptation:** Select the IPC adaptation name from **Section 6.3**.
- **Location:** Select the IPC location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

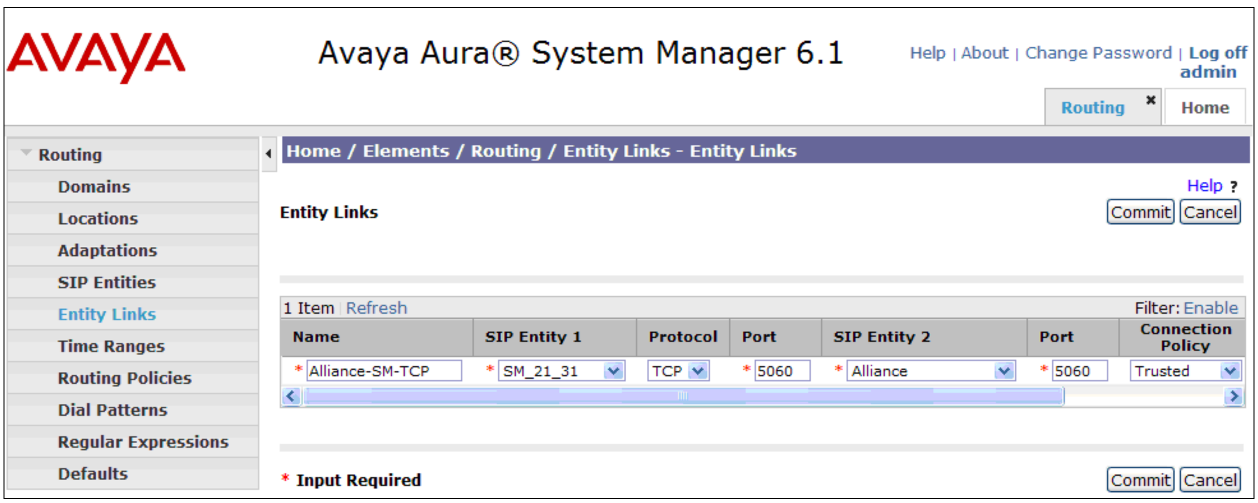


6.5. Administer Entity Links

Select **Routing → Entity Links** from the left pane, and click **New** in the subsequent screen (not shown) to add a new entity link for IPC.

The **Entity Links** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name.
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **SIP Entity 2:** The IPC entity name from **Section 6.4**.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **Connection Policy:** Leave it as Trusted



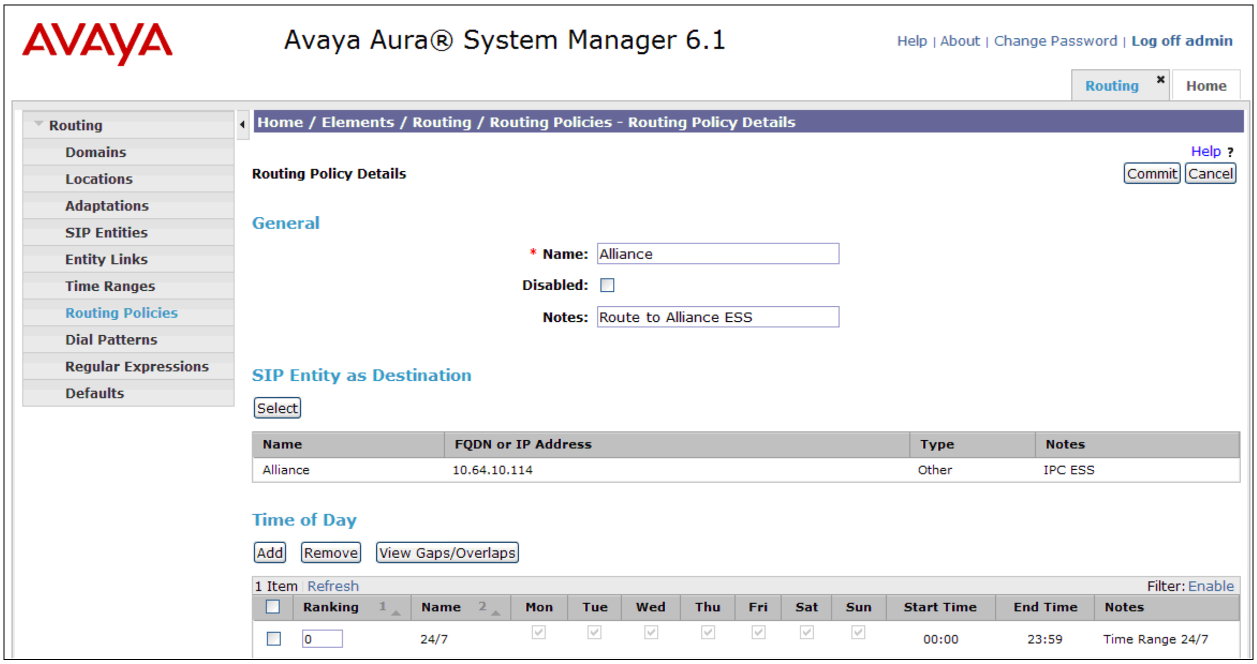
6.6. Administer Routing Policies

Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for IPC.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**.

In the **SIP Entity as Destination** sub-section, click **Select** and select the IPC entity name from **Section 6.4** in the listing (not shown).

Retain the default values in the remaining fields.



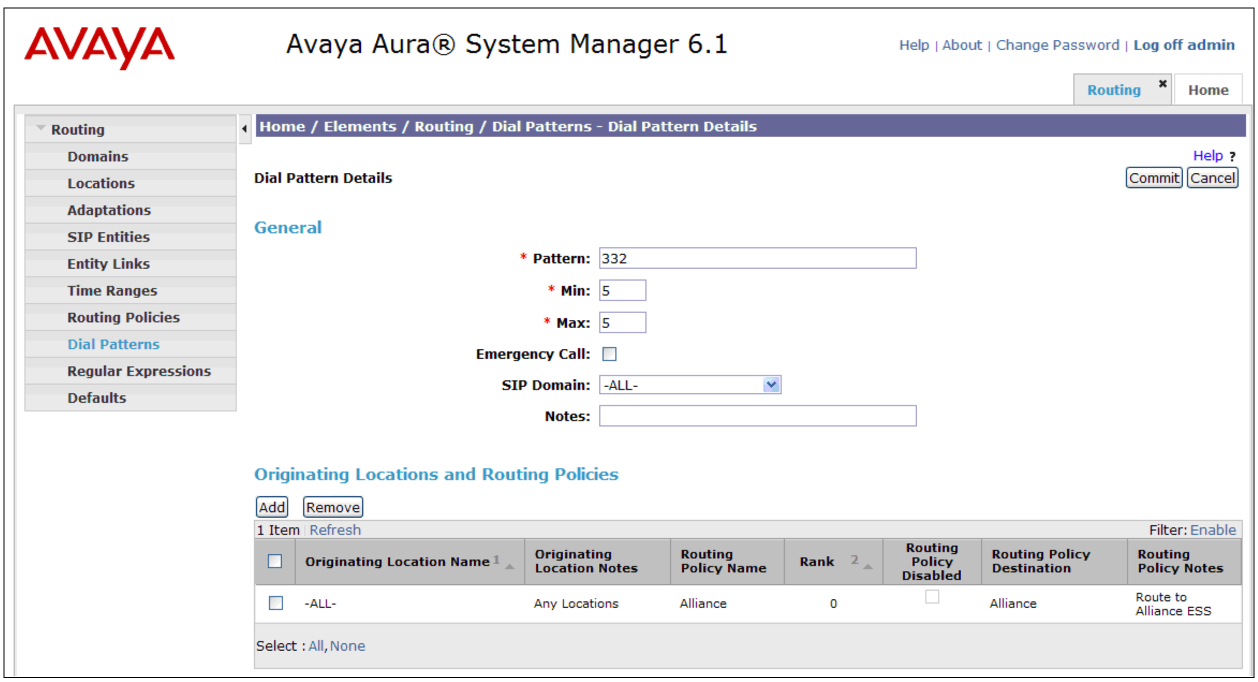
6.7. Administer Dial Patterns

Select **Routing → Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach IPC turret users.

The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match.
- **Min:** The minimum number of digits to be matched.
- **Max:** The maximum number of digits to be matched.
- **SIP Domain:** During the compliance test, "all" was selected for the sip domain.
- **Notes:** Any desired description.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy for reaching IPC turret users. In the compliance testing, the policy allowed for call origination from all locations, as shown below. Retain the default values in the remaining fields.



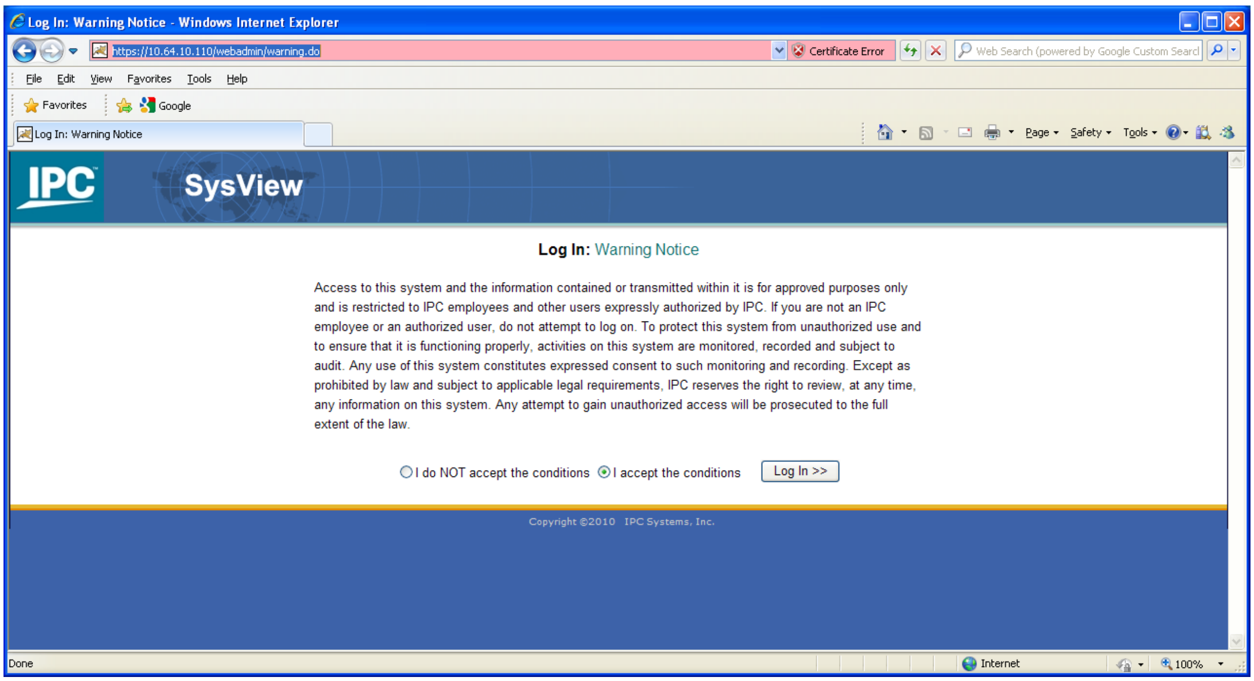
7. Configure IPC System Interconnect

This section provides the procedures for configuring IPC System Interconnect. The procedures include the following areas:

- Configure Route Plan
- Configure SIP Proxy
- Administer Trusted Host
- Configure SIP Trunk

7.1. Configure Route Plan

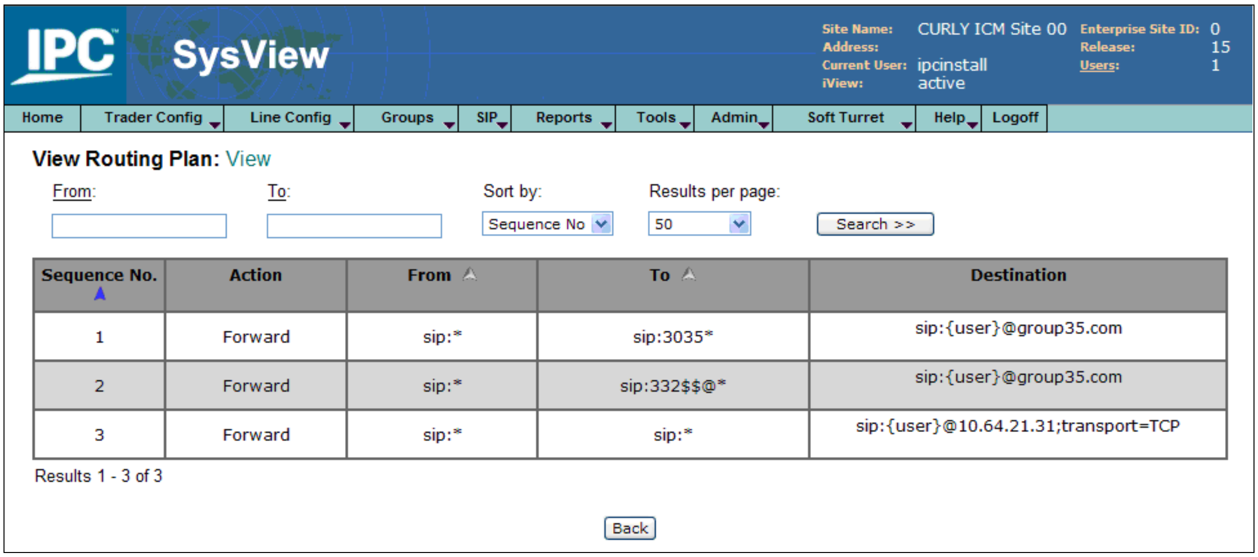
Access the **IPC System Center** web interface by using the URL https://ip-address/webadmin in an Internet browser window, where "ip-address" is the IP address of the System Center. Select **I accept the condition**, and log in using the appropriate credentials.



On the **SysView** page, navigate to **SIP → Routing Plan → View Routing Plan** to view what is used during the compliance test.

The entry with **Sequence No. 2** was used for routing of inbound calls to IPC. Note that the Destination URL contains the internal default value for the SIP trunk card, in this case “group35.com”. The entry with **Sequence No. 3** was used for routing of outbound calls to Session Manager. Note the Destination URL includes the IP address of the signaling interface for Session Manager, and the transport method from **Section 5.4**.

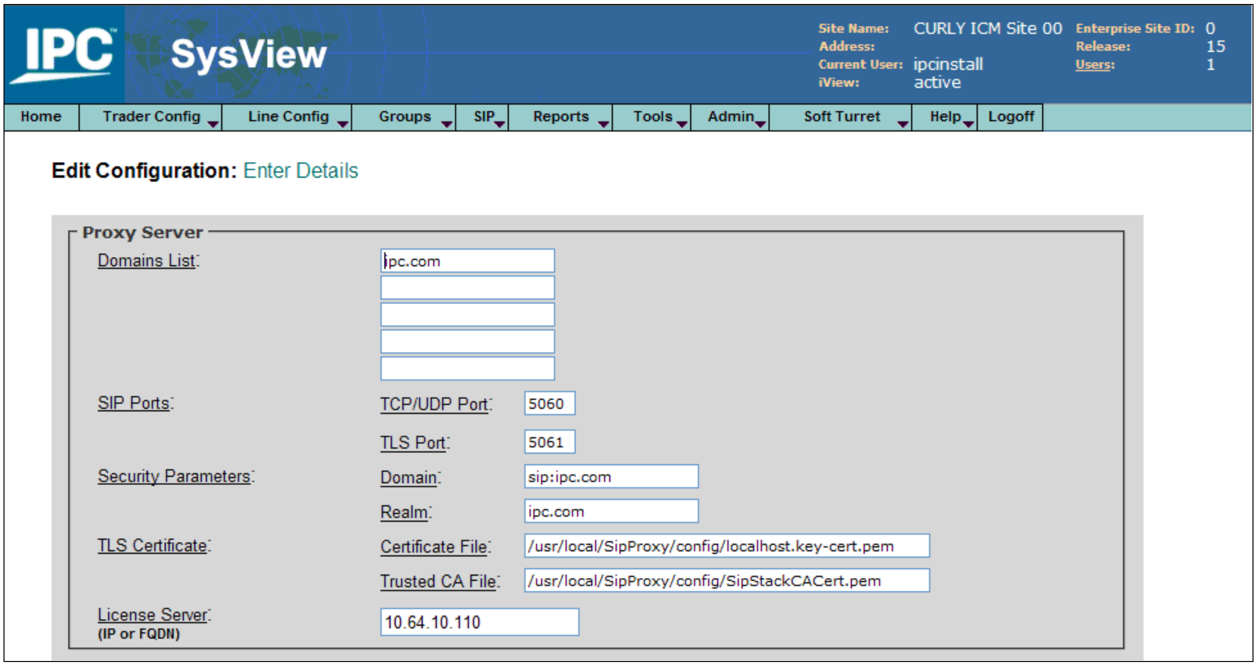
To create a new routing plan, redirect the path to **SIP → Routing Plan → Add Routing Plan**.



Sequence No.	Action	From	To	Destination
1	Forward	sip:*	sip:3035*	sip:{user}@group35.com
2	Forward	sip:*	sip:332$$@*	sip:{user}@group35.com
3	Forward	sip:*	sip:*	sip:{user}@10.64.21.31;transport=TCP

7.2. Configure SIP Proxy

On the **SysView** page, navigate to **SIP → SIP Server → Configuration** to create a new server configuration. Enter a domain that will be used on the IPC side. Provide SIP ports for TCP/UDP and TLS. During the test TCP was used.



7.3. Administer Trusted Host

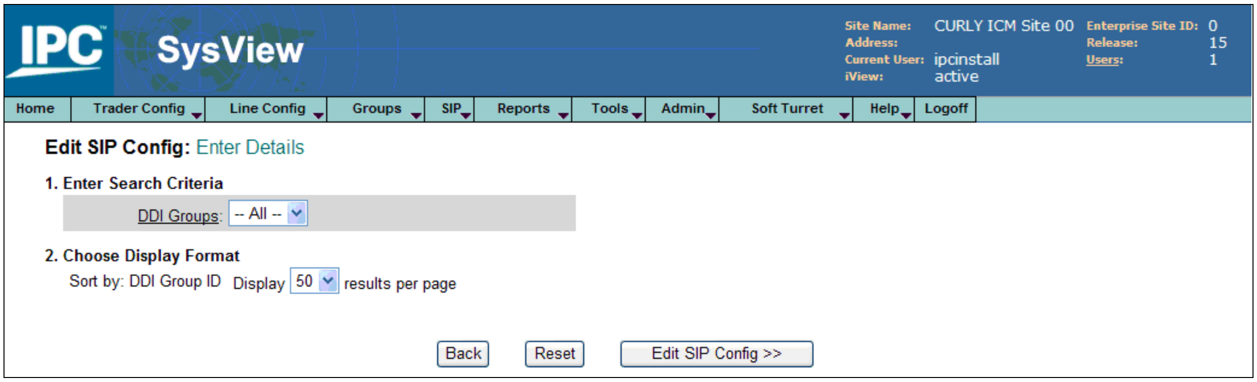
From the Linux shell of the ESS server, navigate to the **/usr/local/SipProxy**/ directory, and issue the command shown below with the "-add" option to add Session Manager as a trusted host. Note that 10.64.21.31 is the IP address of the signaling interface for Session Manager.

The same command can be used with the "-view" option to make certain Session Manager is displayed as a trusted host

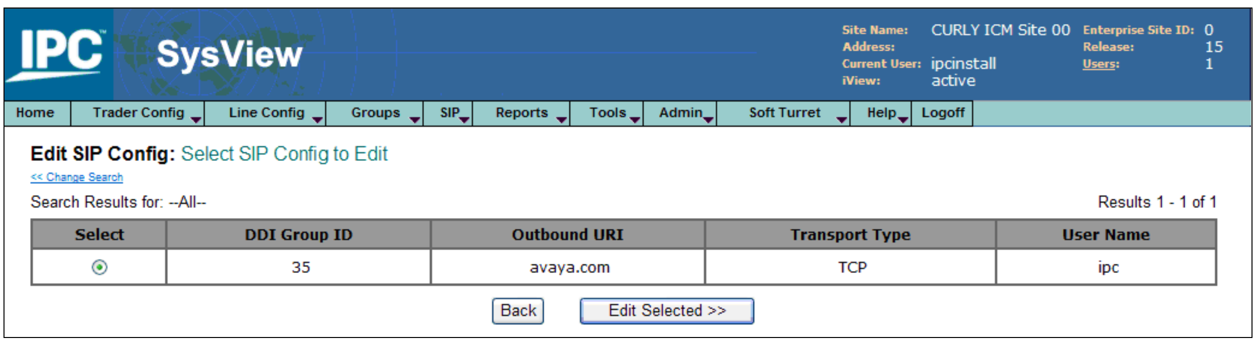
```
[root@esshost ~]# cd /usr/local/SipProxy/
[root@esshost SipProxy]# ./trusted_hosts.pl -add=10.64.21.31

[root@esshost SipProxy]# ./trusted_hosts.pl -view
ip_address      last_modified
10.64.21.31     2012-06-04 15:38:35
```

7.4. Configure SIP Trunk

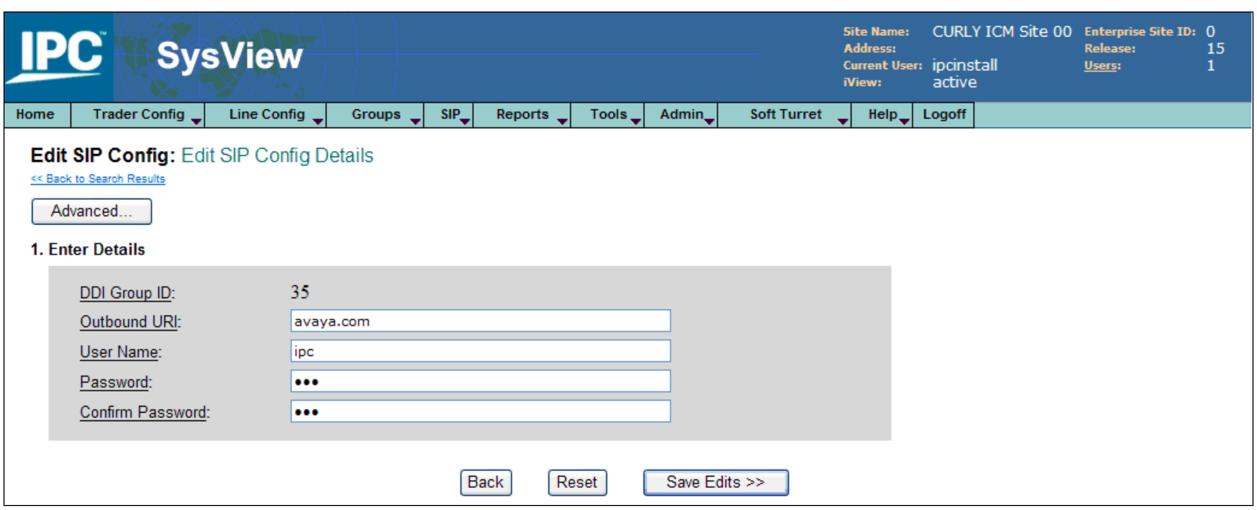
On the **SysView** page, navigate to **SIP → SIP Trunk Parameters** and select the **Edit SIP Config** button.



On the **Select SIP Config to Edit** page, select the relevant SIP **DDI Group ID**, in this case "35" and click on the "Edit Selected" button.



On the **Edit SIP Config Details** page, provide **Outbound URI**.



8. Configure IPC Alliance MX

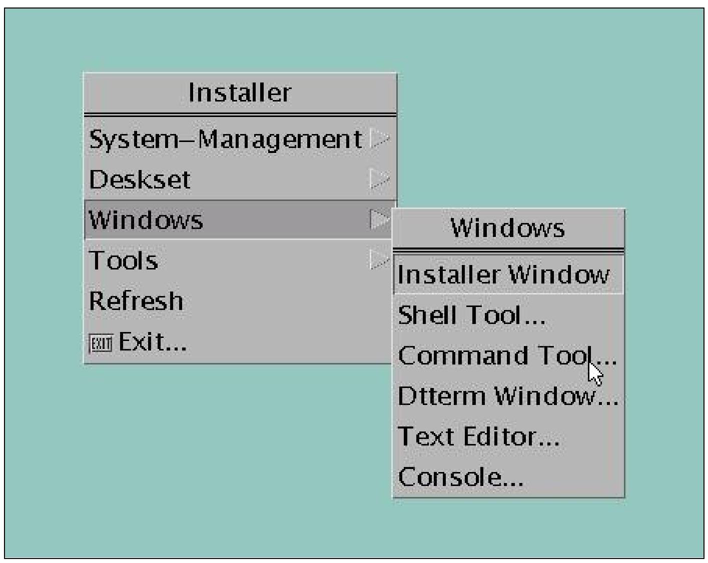
This section provides the procedures for configuring IPC Alliance MX. The procedures include the following areas:

- Launch Iview
- Administer wire groups

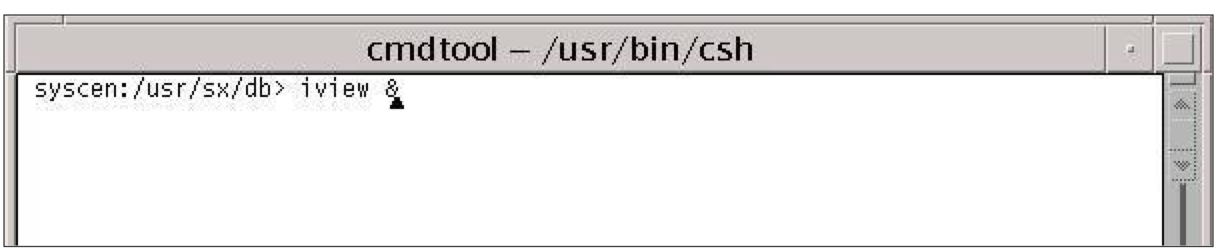
The configuration of Alliance MX is typically performed by IPC installation technicians. The procedural steps are presented in these Application Notes for informational purposes.

8.1. Launch Iview

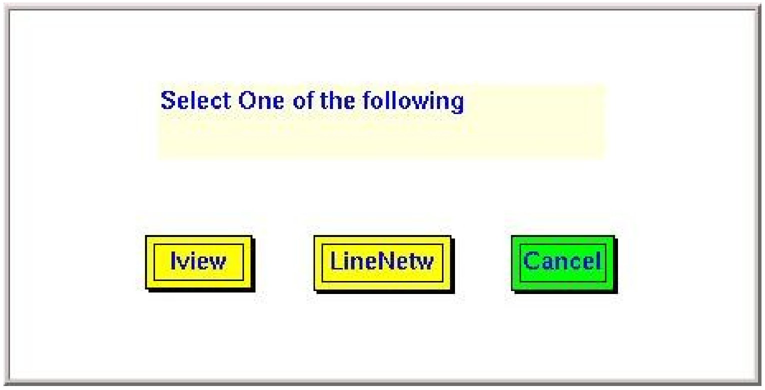
From the Alliance MX console (or System Center console), right-click and select **Windows → Command Tool** from the pop-up boxes.



The **cmdtool** screen is displayed. Enter "iview &", as shown below.

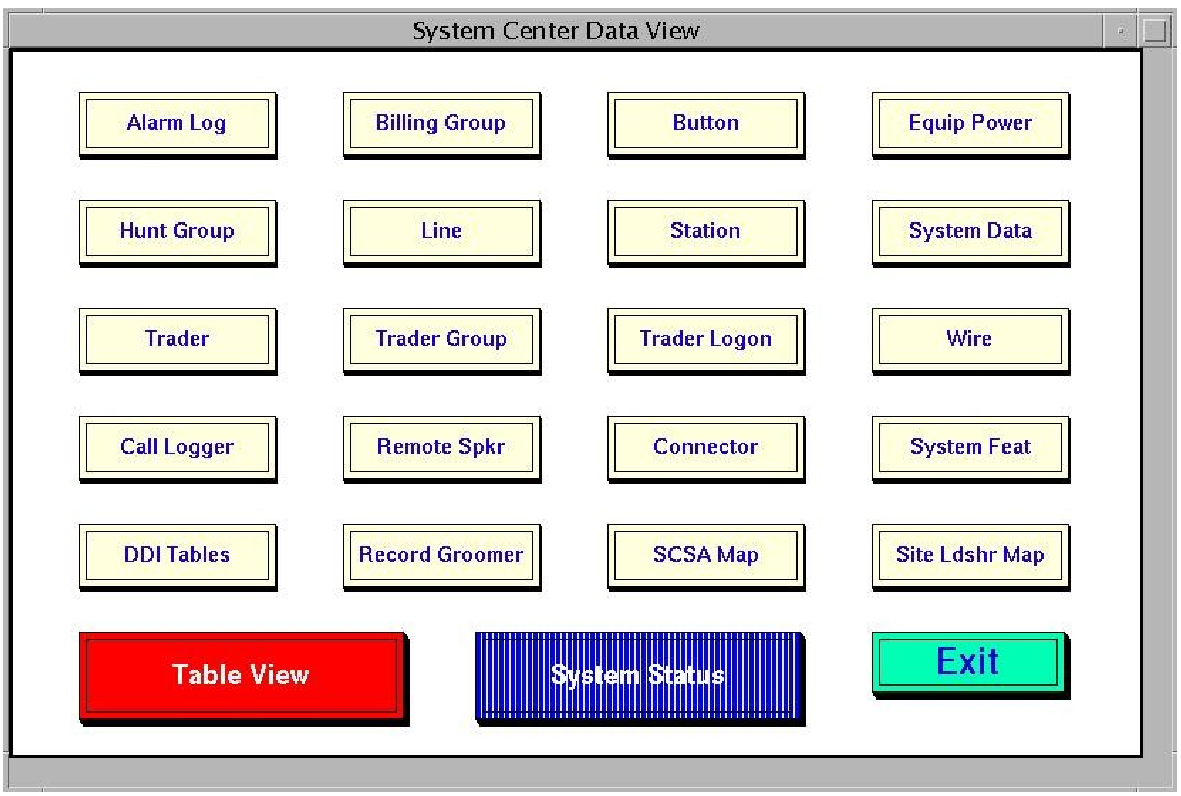


In the pop-up box shown below, click **Iview**.

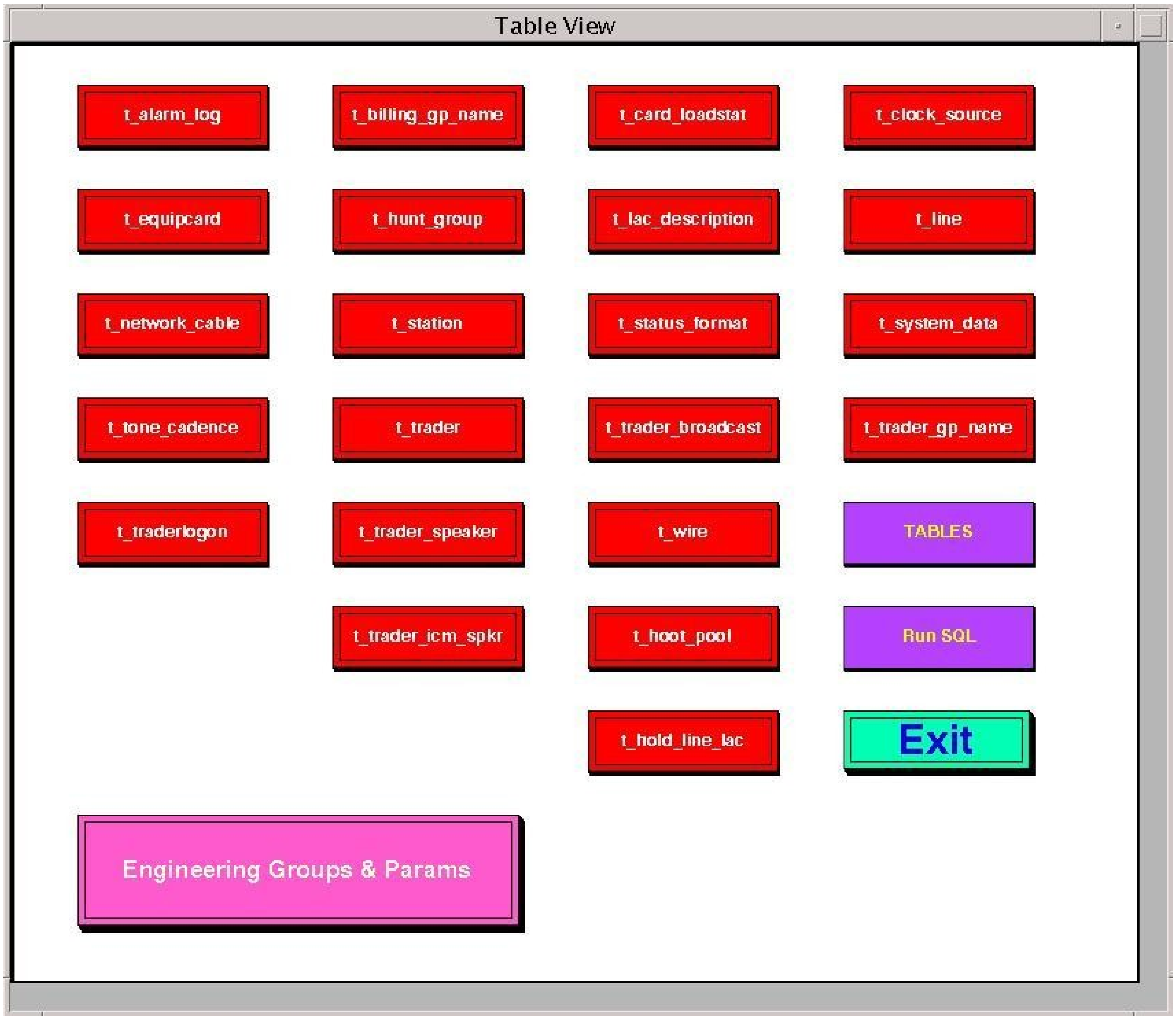


8.2. Administer Wire Groups

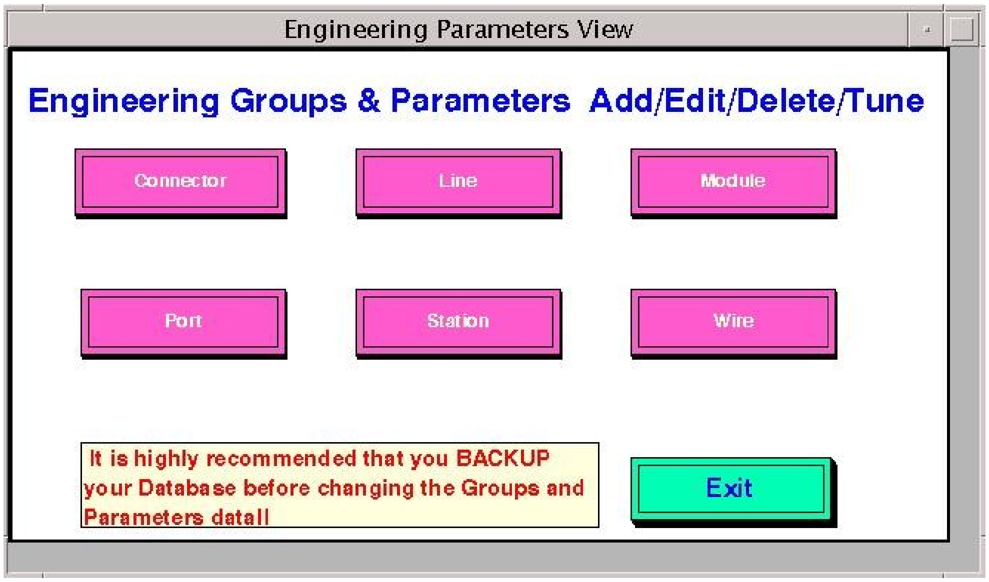
The **System Center Data View** screen is displayed. Click **Table View**.



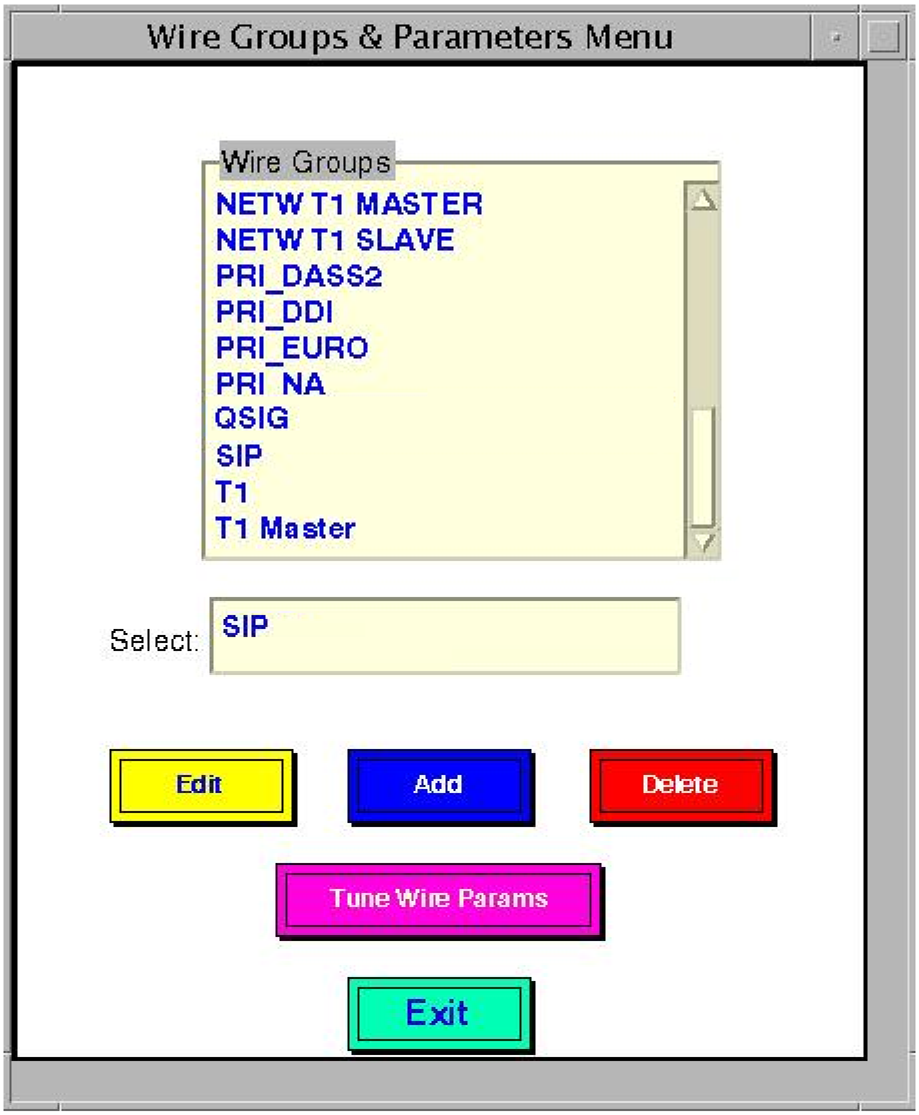
The **Table View** screen is displayed. Click **Engineering Groups & Params**.



The **Engineering Parameters View** screen is displayed next. Click **Wire**.



The **Wire Groups & Parameters Menu** screen is displayed. In the **Wire Groups** sub-section, scroll down and select "SIP". Click **Edit**.



The **p_Wire Edit Group** screen is displayed next. Scroll down the screen as necessary to locate the entry with **Param ID** of "365". Click on the corresponding **New Param Value** field, and enter "2" to denote Avaya as the PBX provider.
Locate the entry with **Param ID** of "370". Click on the corresponding **New Param Value** field, and enter "4" to enable Forward Switching. Scroll down the screen as necessary to locate the entry with **Param ID** of "661". Click on the corresponding **New Param Value** field, and enter "1" to activate detection for G729. Locate the entry with **Param ID** of "666". Click on the corresponding **New Param Value** field, and enter "1" to enable SIP Provisional Acknowledgement (PRACK). Locate the entry with **Param ID** of "668". Click on the corresponding **New Param Value** field, and enter "0" to disable SIP Remote Party ID (RPI).

After the configuration changes, reboot the SIP trunk card or perform a system load.

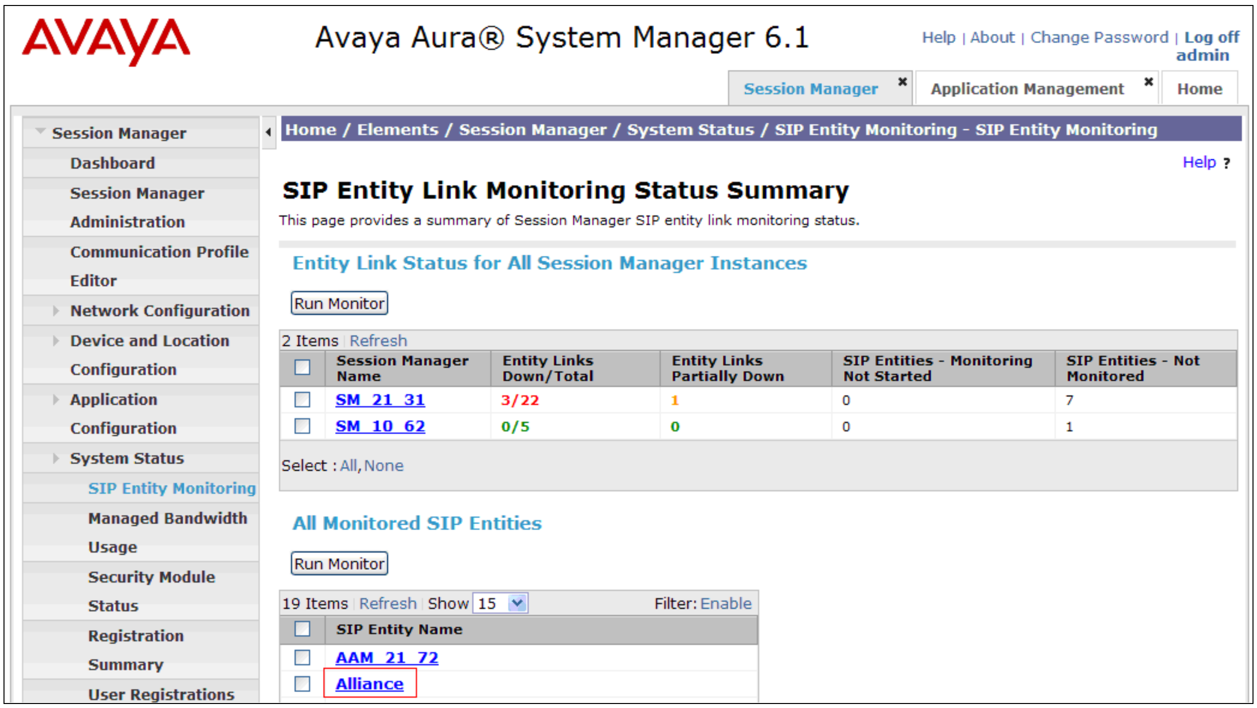
73	SIP Line Card	32767	1	32767	DSP_TERM_ATTEN	DSP TERM threshold	number	141
74	SIP Line Card	0	-5	5	TERM_SHIFT	gain/loss into ipc network	number	362
75	SIP Line Card	0	-5	5	PERIPH_SHIFT	gain/loss into public network	number	363
76	SIP Line Card	6	0	32	INTERDIGIT_TO	interdigit timeout for enbloc signaling	number	364
77	SIP Line Card	2	1	7	PBX_PROVIDER	-7/DEF,AVYA,NRTL,ERISN,MITL,SMNS,CS21	enum	365
78	SIP Line Card	6	1	15	MAX_DIVERTS	Max Number of Diverts per Call	number	369
79	SIP Line Card	4	0	4	FS_ENABLE	0-4/Off, Imm&Busy, RNA, All, Always FS	number	370
80	SIP Line Card	200	200	10000	FS_DELAY	Time(msec) to Wait B4 Forward Switching	number	371
81	SIP Line Card	1	1	5	LN_RECORDS	1-5/NONE,MX_PBX,MWI,DISC,All	number	375
82	SIP Line Card	16	-32767	32767	VPKT CONTROL	Voice Pkt Control	number	642
83	SIP Line Card	10	-32767	32767	VPKT PERIOD	Voice Pkt Period in samples	number	643
84	SIP Line Card	12825	-32767	32767	VPKT JITTERDEPTH	Voice Pkt Jitter Depth in samples	number	644
85	SIP Line Card	0	-32767	32767	VPKT JITTERCTRL	Voice Pkt Jitter Control	number	645
86	SIP Line Card	0	-32767	32767	VPKT SPARE1	Voice Pkt spare1	number	646
87	SIP Line Card	1400	0	3000	INTRUSION_FREQ	Intrusion frequency, Hz	number	647
88	SIP Line Card	350	0	3000	DIALTONELO_FREQ	Dialtone LO frequency, Hz	number	648
89	SIP Line Card	440	0	3000	DIALTONEHI_FREQ	Dialtone HI frequency, Hz	number	649
90	SIP Line Card	480	0	3000	BUSYTONELO_FREQ	Busytone LO frequency, Hz	number	650
91	SIP Line Card	620	0	3000	BUSYTONEHI_FREQ	Busytone HI frequency, Hz	number	651
92	SIP Line Card	440	0	3000	RINGBACKLO_FREQ	Ringback LO frequency, Hz	number	652
93	SIP Line Card	480	0	3000	RINGBACKHI_FREQ	Ringback HI frequency, Hz	number	653
94	SIP Line Card	480	0	3000	ERRTONELO_FREQ	Error tone LO frequency, Hz	number	654
95	SIP Line Card	620	0	3000	ERRTONEHI_FREQ	Error tone HI frequency, Hz	number	655
96	SIP Line Card	1209	0	3000	SPLSHTONELO_FREQ	Splash tone LO frequency, Hz	number	656
97	SIP Line Card	1477	0	3000	SPLSHTONEHI_FREQ	Splash tone HI frequency, Hz	number	657
98	SIP Line Card	1400	0	3000	RECWARNTONE_FREQ	Record warning frequency, Hz	number	658
99	SIP Line Card	0	0	10000	MRD Ringback Ton	Ringback Tone Duration (msec)	number	659
100	SIP Line Card	1	0	1	VAD	Voice Activity Detection	number	661
101	SIP Line Card	0	0	1	MWI Subscribe	Send MWI Subscribe, Off = 0, On = 1	number	663
102	SIP Line Card	0	0	1	SIP Divert	HistoryInfo = 0, CCDiversion = 1	number	664
103	SIP Line Card	1	0	1	SIP PRACK	Enable SIP Provisional ACK	number	666
104	SIP Line Card	1	0	1	SIP PAI	Enable SIP P-Asserted Identity	number	667
105	SIP Line Card	0	0	1	SIP RPID	Enable SIP Remote Party ID	number	668
106	SIP Line Card	0	0	1	AEC_Enable	Enable AEC Control Filter	number	669
107	SIP Line Card	0	-3	3	AEC_Control	AEC Aggression level	number	670
108	SIP Line Card	0	0	1	AEC_NR_Filter	Enable AEC Noise Reduction	number	671
109	SIP Line Card	1	0	1	VoIP Stat Log	Enable VoIP Statistics Logging	number	672
110	SIP Line Card	1	0	1	SIP 3264 Hold	Enable SIP 3264 Call Hold/Resume	number	673
111	SIP Line Card	1	0	1	SIP Conn Party U	Enable SIP connected party update messag	number	674
112	SIP Line Card	15	0	15	FRF11 Idle Signa	FRF11 Idle bit pattern	number	675
113	SIP Line Card	10	0	15	FRF11 Seize Sign	FRF11 Seize bit pattern	number	676
114								

9. Verification Steps

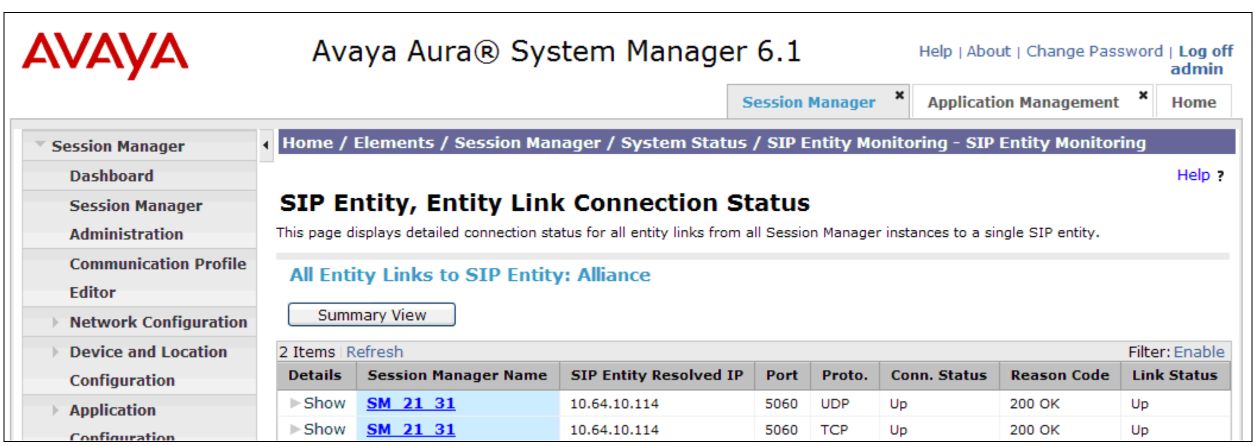
This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Session Manager and IPC Alliance MX.

9.1. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements → Session Manager** to display the **Session Manager Dashboard** screen (not shown). Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click on the IPC entity name from **Section 6.4**.

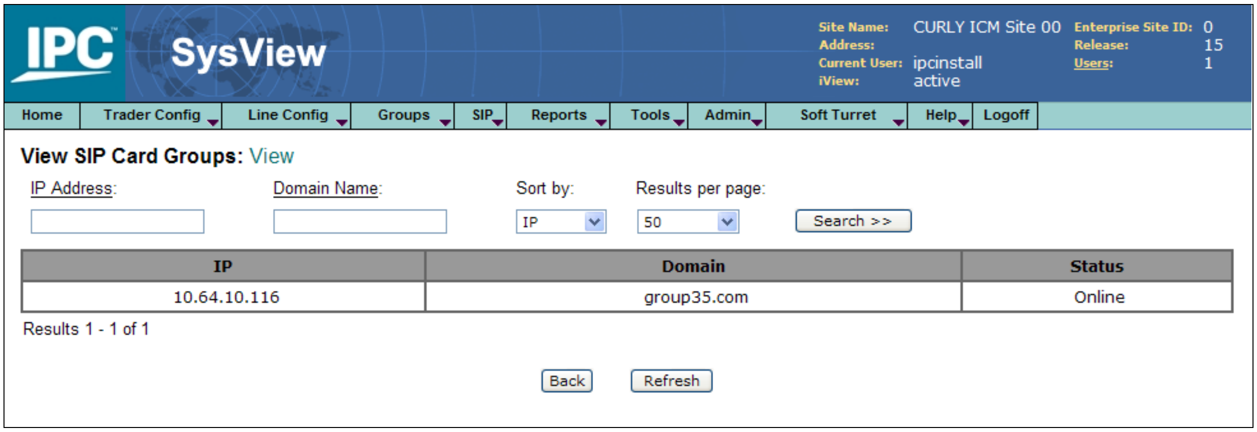


The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that **Conn. Status** and **Link Status** are "Up", as shown below.



9.2. Verify IPC System Interconnect

From the SysView web interface, select **SIP → Update ESS with SIP Trunk Info → View SIP Cards Groups**. Verify that there is an entry that corresponds to SIP card number. Verify that the **Status** is "Online", as shown below.



10. Conclusion

These Application Notes describe the configuration steps required for IPC Alliance MX 15.03 to successfully interoperate with Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager 6.1 using SIP trunks to Avaya Aura® Session Manager. All feature and serviceability test cases were completed.

11. Additional References

This section references the product documentation relevant to these Application Notes.

- *Administering Avaya Aura™ Communication Manager*, Document 03-300509, Issue 6.0, Release 6.0, June 2010, available at http://support.avaya.com.

- *IPC PATCH 15.03.00.07a Intall Guide*, Revision Number 7, April 2011, available upon request to IPC Support.

- *Nexus Suite 2.0 SP1 Patch11 or Higher Deployment Guide*, Part Number B02200161, Revision Number 01, available upon request to IPC Support.

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.