**Avaya Solution & Interoperability Test Lab**

# Application Notes for IPC System Interconnect with Avaya IP Office – Issue 1.0

## Abstract


These Application Notes describe the configuration steps required for IPC System Interconnect to interoperate with Avaya IP Office.

IPC System Interconnect is a trading communication solution. In the compliance testing, IPC System Interconnect used SIP trunks to Avaya IP Office, for turret users on IPC to reach users on Avaya IP Office and on the PSTN.

The Avaya IP Office Voicemail Pro was used in the test configuration to provide voicemail service for the Avaya IP Office users. The IPC turret users do not have any voicemail capabilities in the test configuration.

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 to interoperate with Avaya IP Office.

IPC System Interconnect is a trading communication solution. In the compliance testing, IPC System Interconnect used SIP trunks to Avaya IP Office, for turret users on IPC to reach users on Avaya IP Office and on the PSTN.

This configuration focused on SIP interoperability between IPC System Interconnect and Avaya IP Office. Avaya IP Office did not provide voicemail service for the IPC turret users in this configuration.

# 2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established among IPC turret users with Avaya IP Office 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, hold/reconnect, DTMF, call forwarding unconditional/ring-no-answer/busy, blind/attended transfer, attended and conference. In addition, voicemail coverage for the Avaya IP Office users was also included.

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 the IPC ESS server.

## 2.2. Test Results

All test cases were executed. The following were the observations from the compliance testing.

- For calls from IPC to Avaya, the IPC turret users will only see the number of the Avaya IP Office user in the display without the name.

- Avaya IP Office users cannot complete an attended transfer of a call from IPC back out to IPC. The workaround is to use blind transfer, or to use conference followed by subsequent drop of the conference-from party.

## 2.3. Support

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

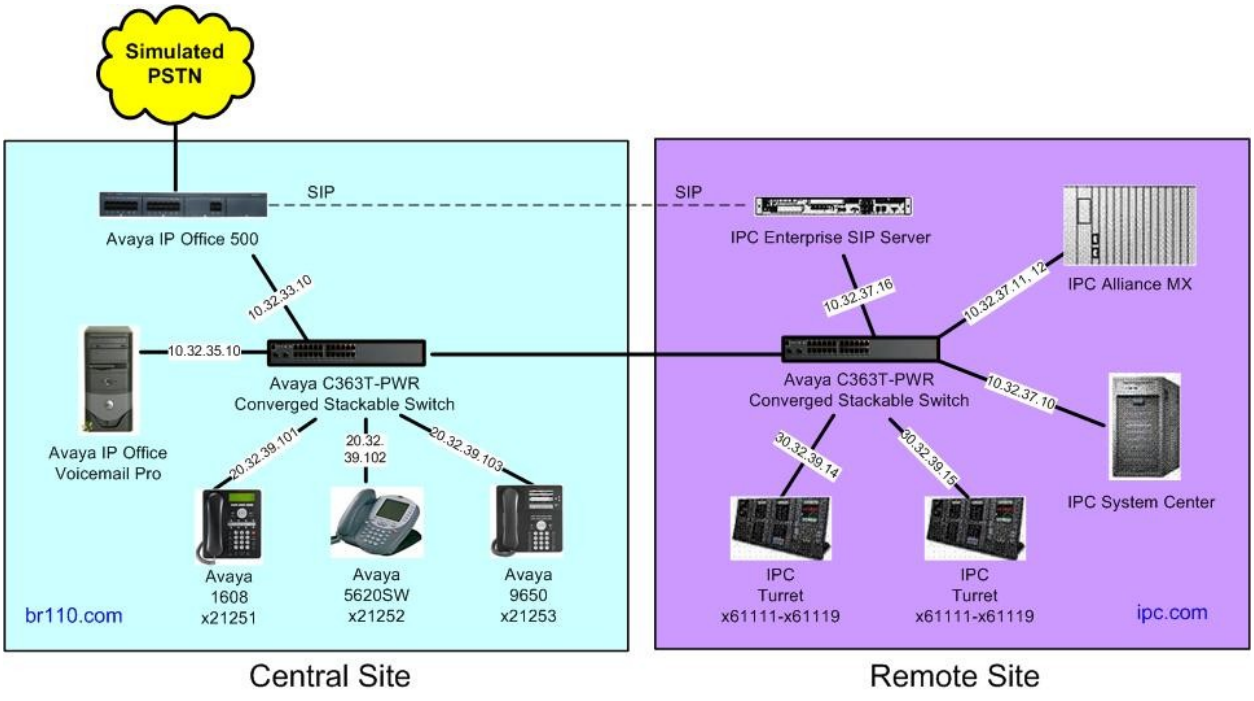
- **Phone:** (800) NEEDIPC, (203) 339-7800
- **Email:** systems.support@ipc.com

# 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 were used from System Interconnect to Avaya IP Office, to reach users on Avaya IP Office and on the PSTN.

The Avaya IP Office Voicemail Pro was used in the test configuration to provide voicemail service for the Avaya IP Office users. The IPC turret users do not have any voicemail capabilities in the test configuration.

A five digit dial plan was used to facilitate dialing between the Central and Remote sites. Unique extension ranges were associated with Avaya IP Office users at the Central site (21xxx), and IPC turret users at the Remote site (61xxx).



# 4. Equipment and Software Validated

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

| Equipment | Software |
|---|---|
| Avaya IP Office 500 | 7.0 (5) |
| Avaya 1608 IP Telephone (H.323) | 1.3 |
| Avaya 5620SW IP Telephone (H.323) | 2.9010 |
| Avaya 9650 IP Telephone (H.323) | 3.1000 |
| IPC System Interconnect<br>• Alliance MX<br>• Enterprise SIP Server<br>• System Center<br>o SIPX Line Card<br>• Turrets | SipProxy-2.00.01-13<br>16.01.01.04.0005<br>16.01.01.04.0005<br>16.01.01.04.0005<br>16.01.01.04.0005<br>16.01.01.04.0005 |

# 5. Configure Avaya IP Office

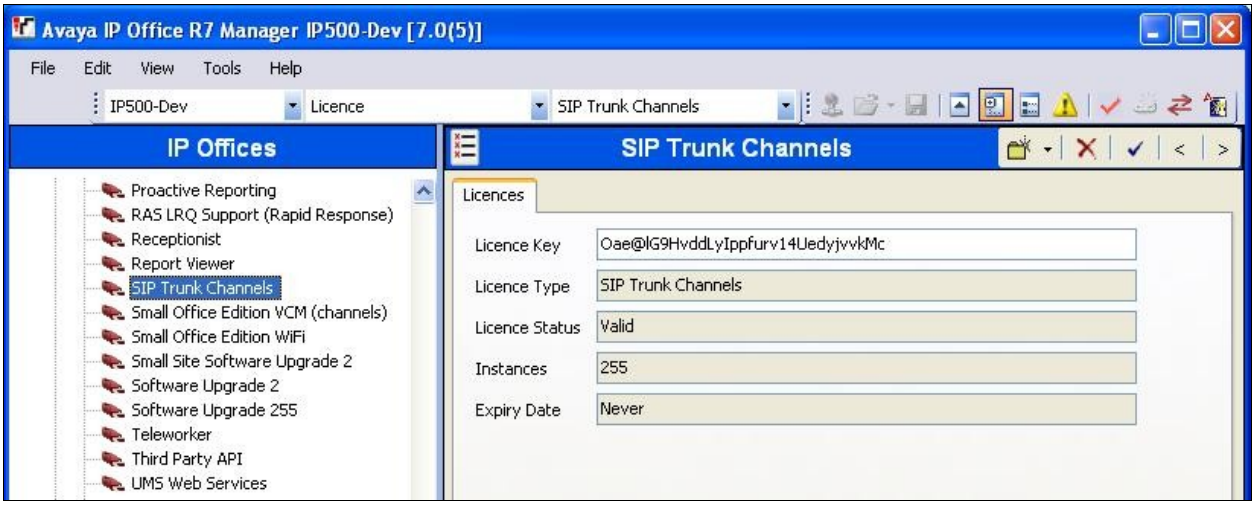
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Verify IP Office license
- Obtain LAN IP address
- Enable SIP trunks
- Administer SIP line
- Administer incoming call route
- Administer short code
- Administer users

## 5.1. Verify IP Office License

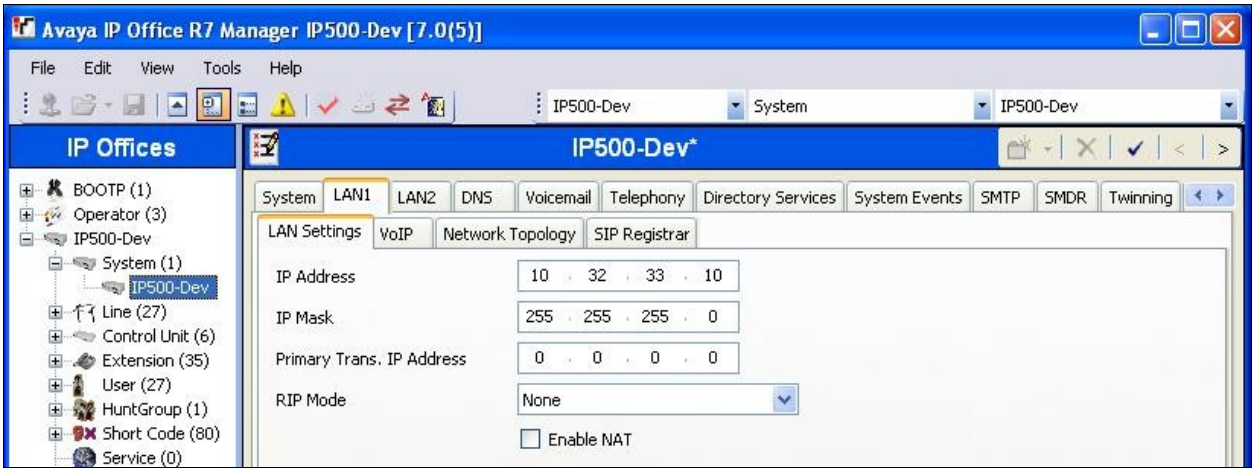
From a PC running the Avaya IP Office Manager application, select **Start > Programs > IP Office > Manager** to launch the Manager application. Select the proper IP Office system, and log in with the appropriate credentials (not shown).

The **Avaya IP Office R7 Manager** screen is displayed. From the configuration tree in the left pane, select **Licence > SIP Trunk Channels** to display the **SIP Trunk Channels** screen in the right pane. Verify that the **Licence Status** is "Valid", and that the **Instances** value is sufficient for the desired maximum number of simultaneous SIP trunk channels.



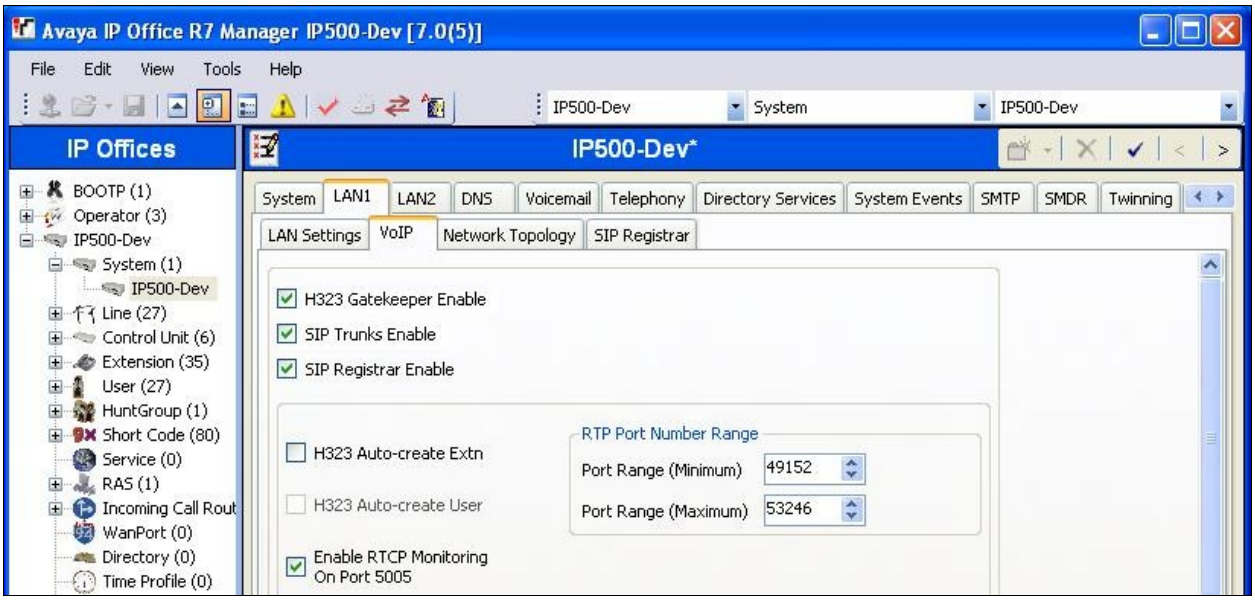
## 5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **IP500-Dev** screen in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure IPC. Note that IP Office can support SIP trunks on the LAN1 and/or LAN2 interfaces, and the compliance testing used the LAN1 interface.




## 5.3. Enable SIP Trunks

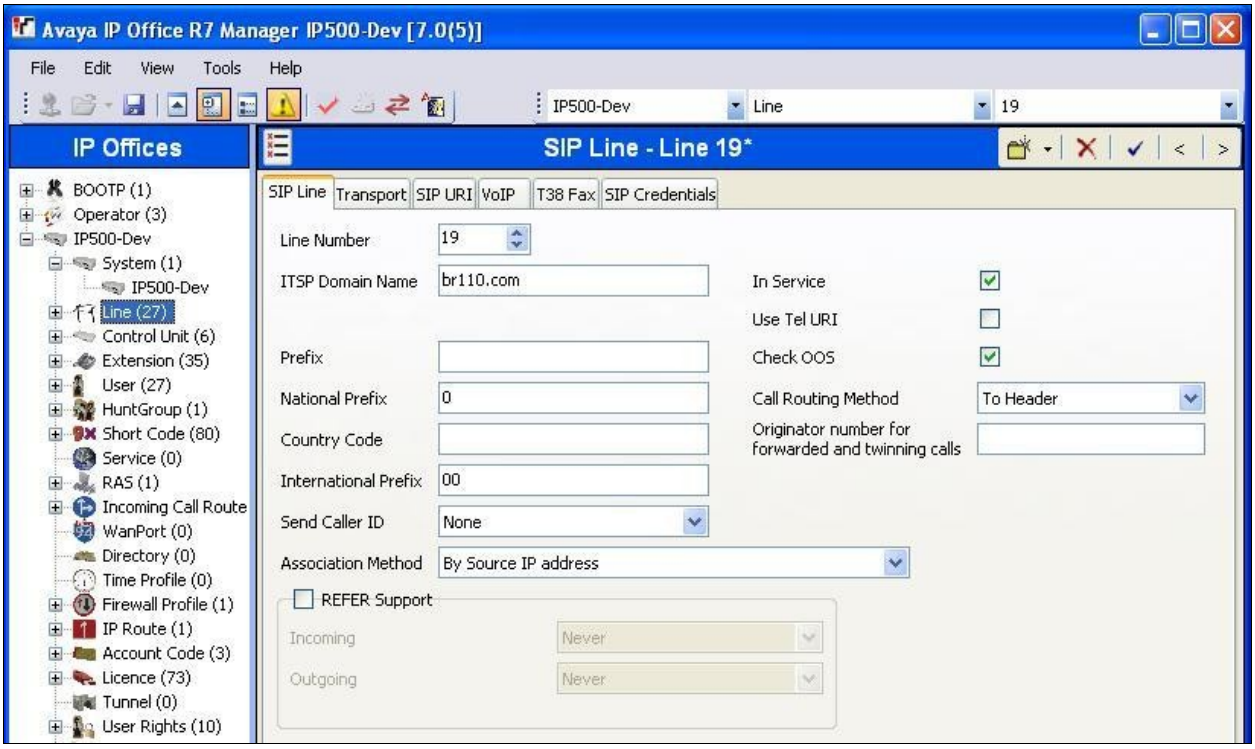
Select the **VoIP** sub-tab. Make certain that **SIP Trunks Enable** is checked, as shown below.



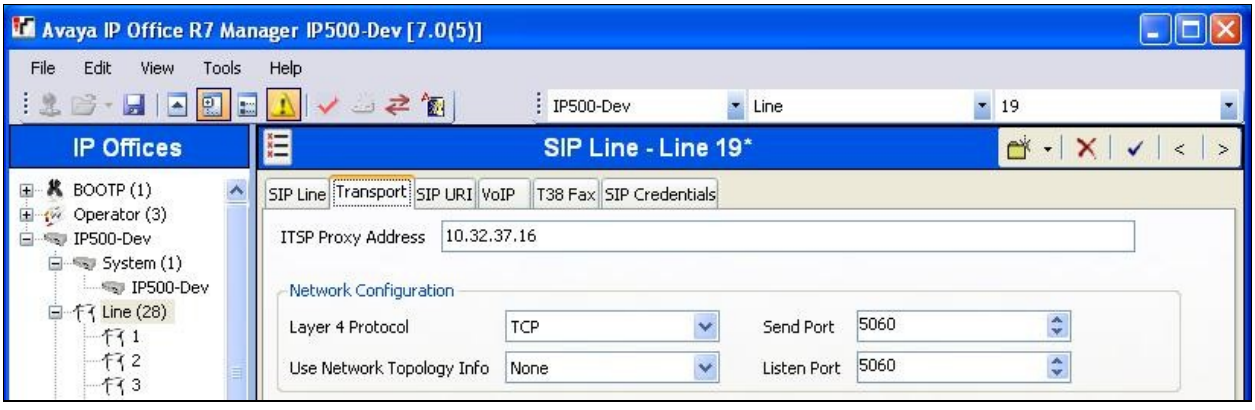
## 5.4. Administer SIP Line

From the configuration tree in the left pane, right-click on **Line**, and select **New > SIP Line** from the pop-up list to add a new SIP line.

The **SIP Line** tab is displayed. For **ITSP Domain Name**, enter the applicable domain name for the network configuration, in this case "br110.com". Uncheck **REFER Support**, as shown below. Retain the default values in the remaining fields.

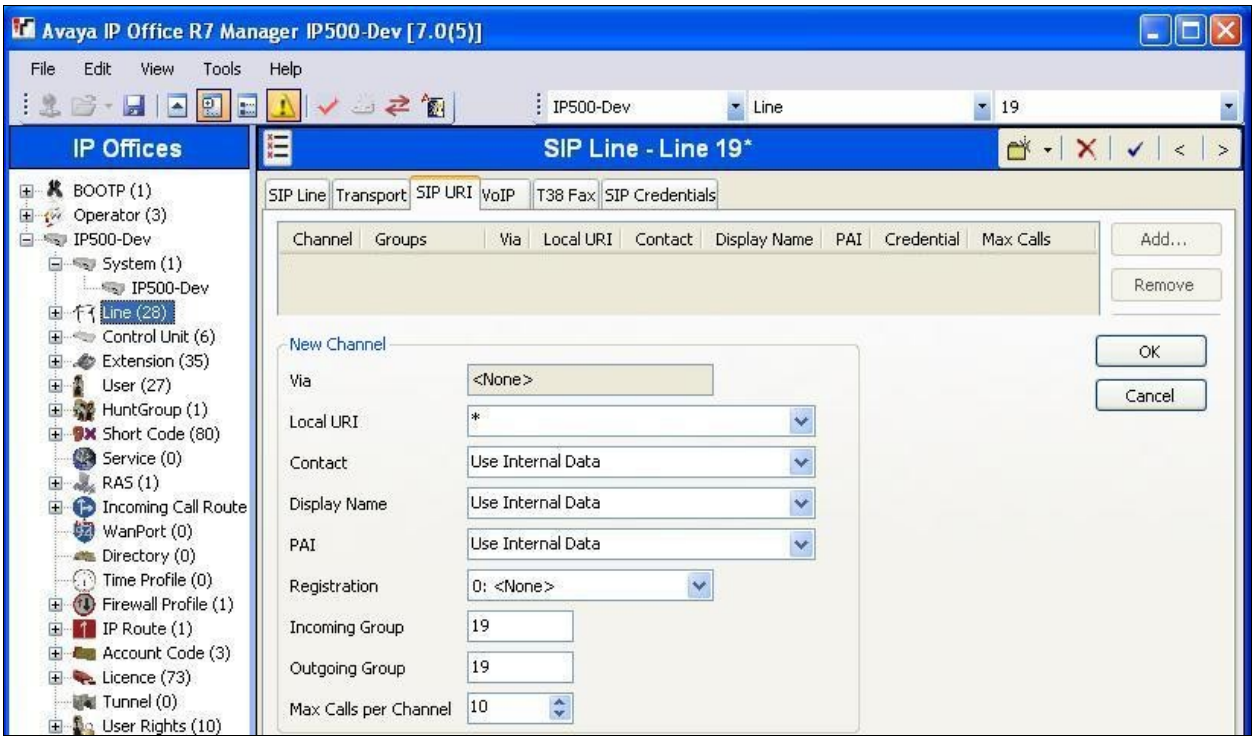



Select the **Transport** tab in the right pane. For **ITSP Proxy Address**, enter the IP address of the IPC ESS server. For **Layer 4 Protocol**, select "TCP". Retain the default values for the remaining fields.

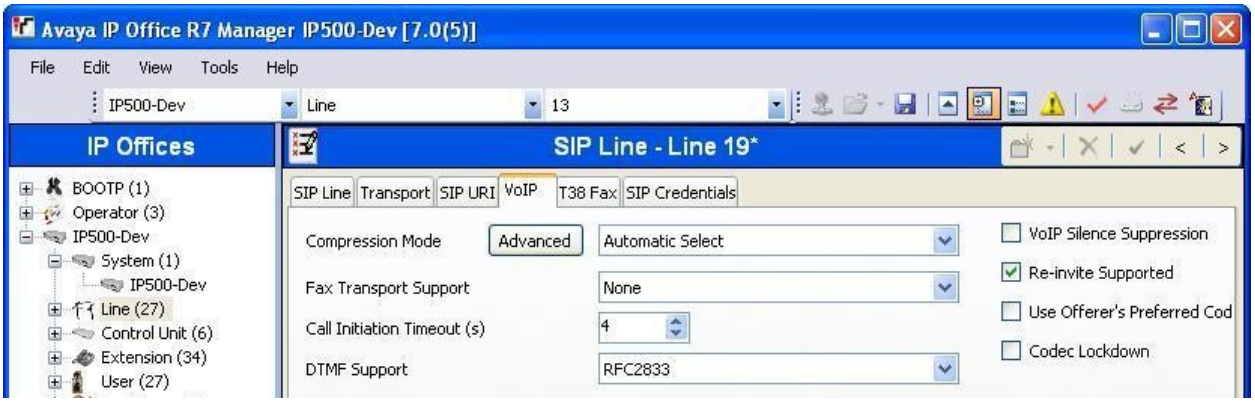


Select the **SIP URI** tab, and click **Add** to display the **New Channel** section. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Local URI:** Enter the wildcard character "*".
- **Contact:** "Use Internal Data"
- **Display Name:** "Use Internal Data"
- **PAI:** "Use Internal Data"
- **Incoming Group:** An unused group number.
- **Outgoing Group:** An unused group number.
- **Max Calls per Channel:** The desired maximum number of simultaneous calls.

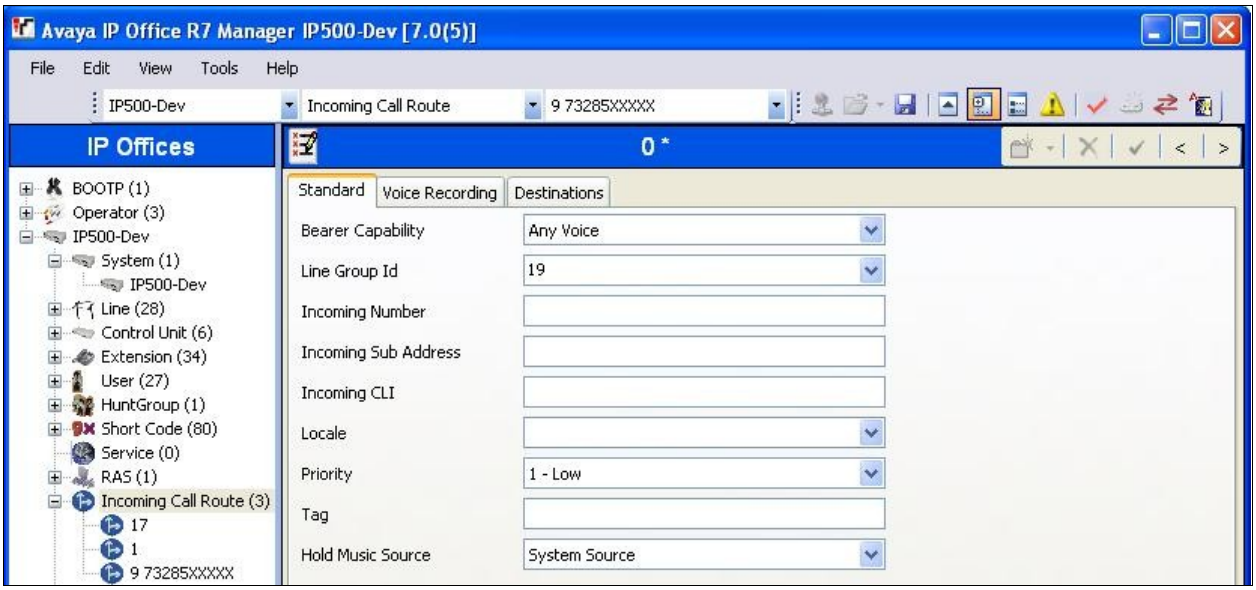



Select the **VoIP** tab, and check **Re-invite Supported**. Retain the default values for the remaining fields.

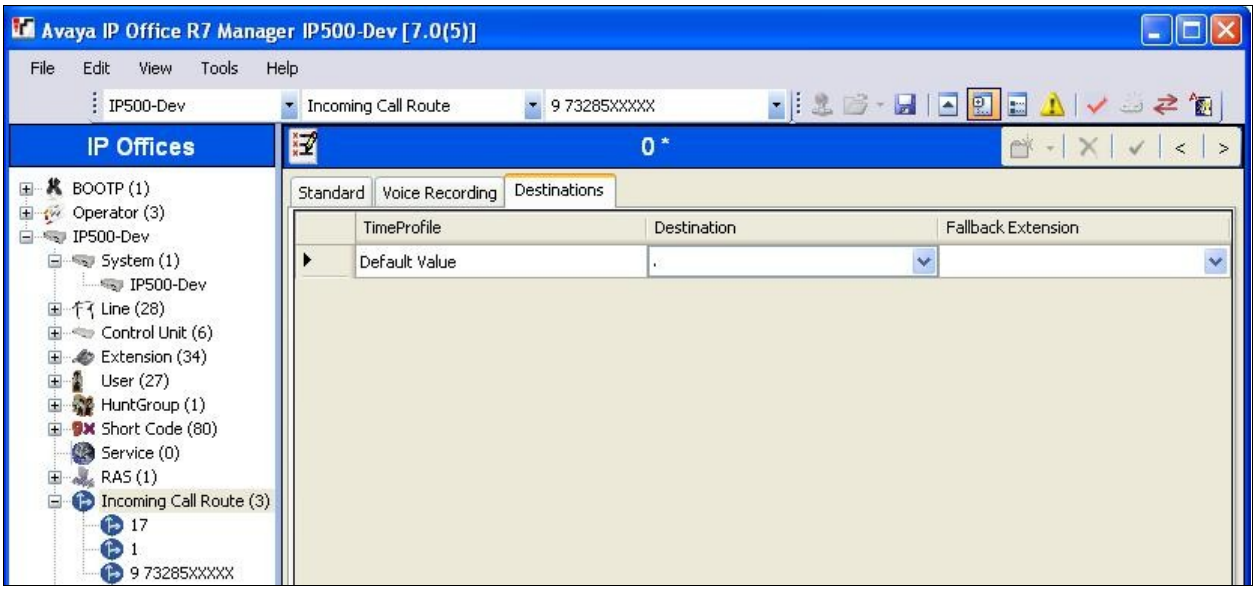


## 5.5. Administer Incoming Call Route

From the configuration tree in the left pane, right-click on **Incoming Call Route**, and select **New** from the pop-up list to add a new route. For **Line Group Id**, select the incoming group number from **Section 5.4**, in this case "19".



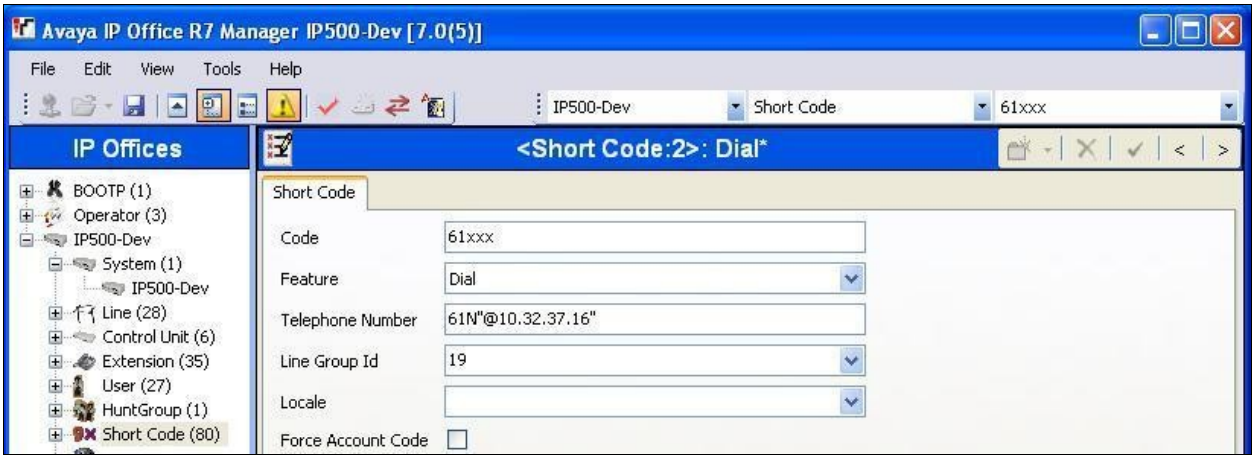

Select the **Destinations** tab. For **Destination**, enter "." to match any dialed number from IPC.



## 5.6. Administer Short Code

From the configuration tree in the left pane, right-click on **Short Code** and select **New** from the pop-up list to add a new short code for calls to IPC. In the compliance testing, users on IPC are designated with extensions 61xxx, and the calls are routed over the SIP trunk to the IPC ESS server.

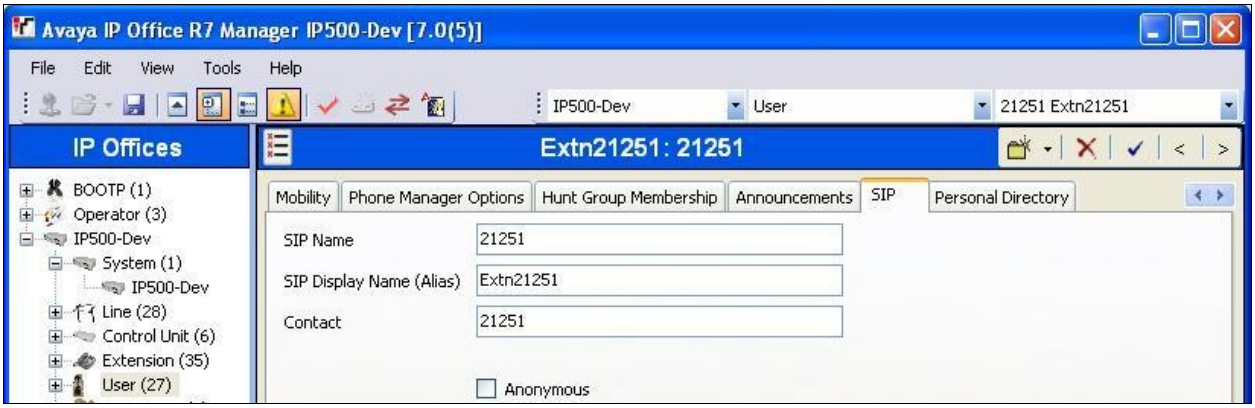
For **Code**, enter "61xxx". For **Feature**, select "Dial" from the drop-down list. For **Telephone Number**, enter the value shown below where "61N" is for the dialed extension and "10.32.37.16" is the IP address of the IPC ESS server. For **Line Group Id**, enter the outgoing group number from **Section 5.4**.




## 5.7. Administer Users

From the configuration tree in the left pane, select the first user from **Section 3** that will be placing and receiving calls via the SIP trunks with IPC. In this case, the user is "21251". Navigate to the **SIP** tab. For **SIP Name**, **SIP Display Name**, and **Contact**, enter the desired values to be used in the SIP URI's **From**, **Display Name**, and **Contact** fields respectively.

Repeat this section for all users placing and receiving calls with IPC. In the compliance testing, three users with extensions 21251-21253 were configured.



# 6. Configure IPC System Interconnect

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

- Launch One Management System
- Administer SIP configuration
- Administer routing plan
- Administer wire groups
- Administer trusted host

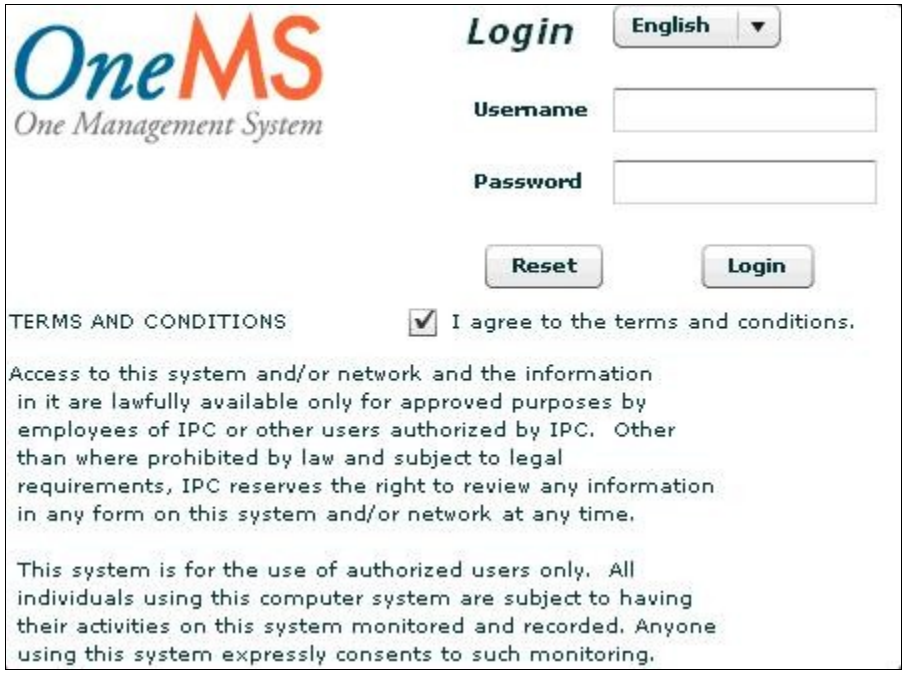
The configuration of System Interconnect is typically performed by IPC installation technicians. The procedural steps are presented in these Application Notes for informational purposes.

## 6.1. Launch One Management System

Access the One Management System web interface by using the URL "http://<ip-address>/oneview" in an Internet browser window, where <ip-address> is the IP address of IPC System Center. Log in using the appropriate credentials.

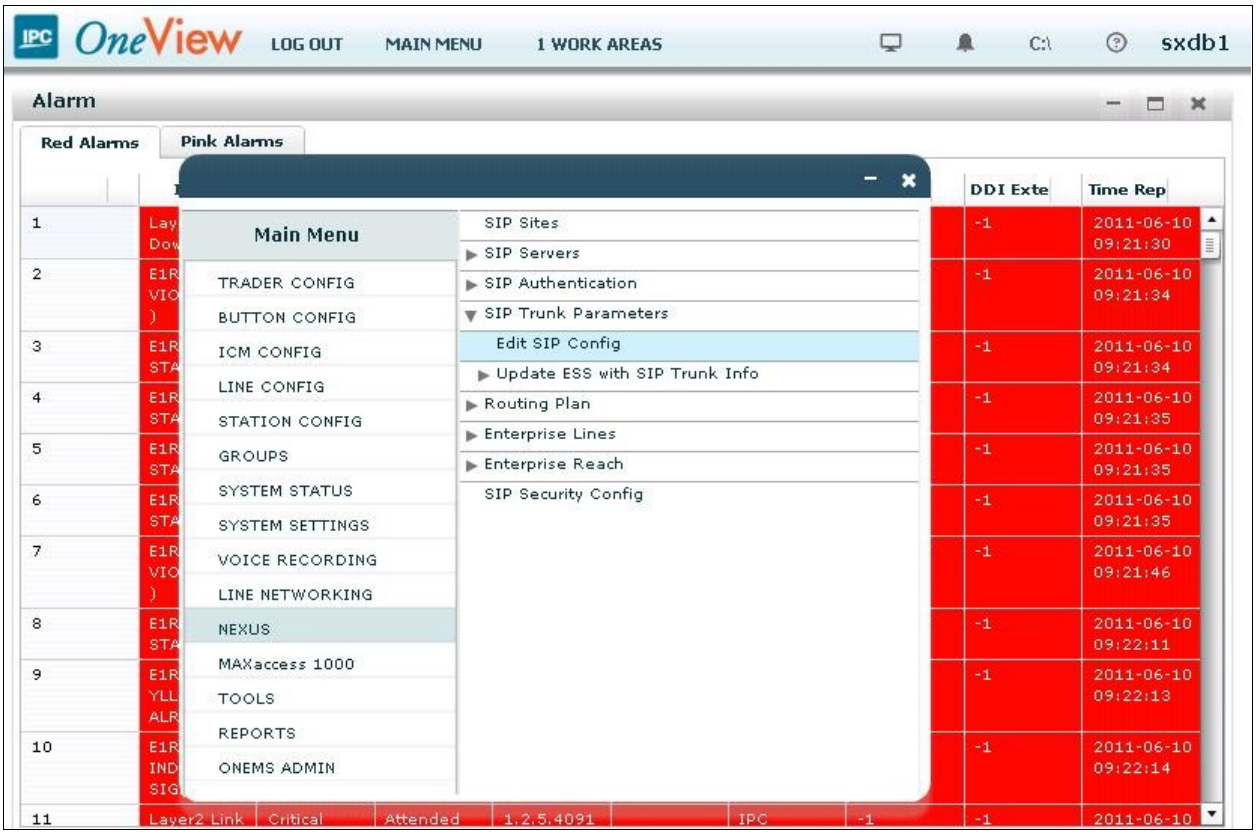
The **Login** screen is displayed. Enter the appropriate credentials. Check **I agree to the terms and conditions**, and click **Login**.

The **License Login** screen is displayed next (not shown). Enter the appropriate password and click **Login**. In the subsequent **Login Information** screen (not shown), click **Continue**.



## 6.2. Administer SIP Configuration

The screen below is displayed next, with the **Main Menu** screen in the forefront. Select **NEXUS > SIP Trunk Parameters > Edit SIP Config**, as shown below.

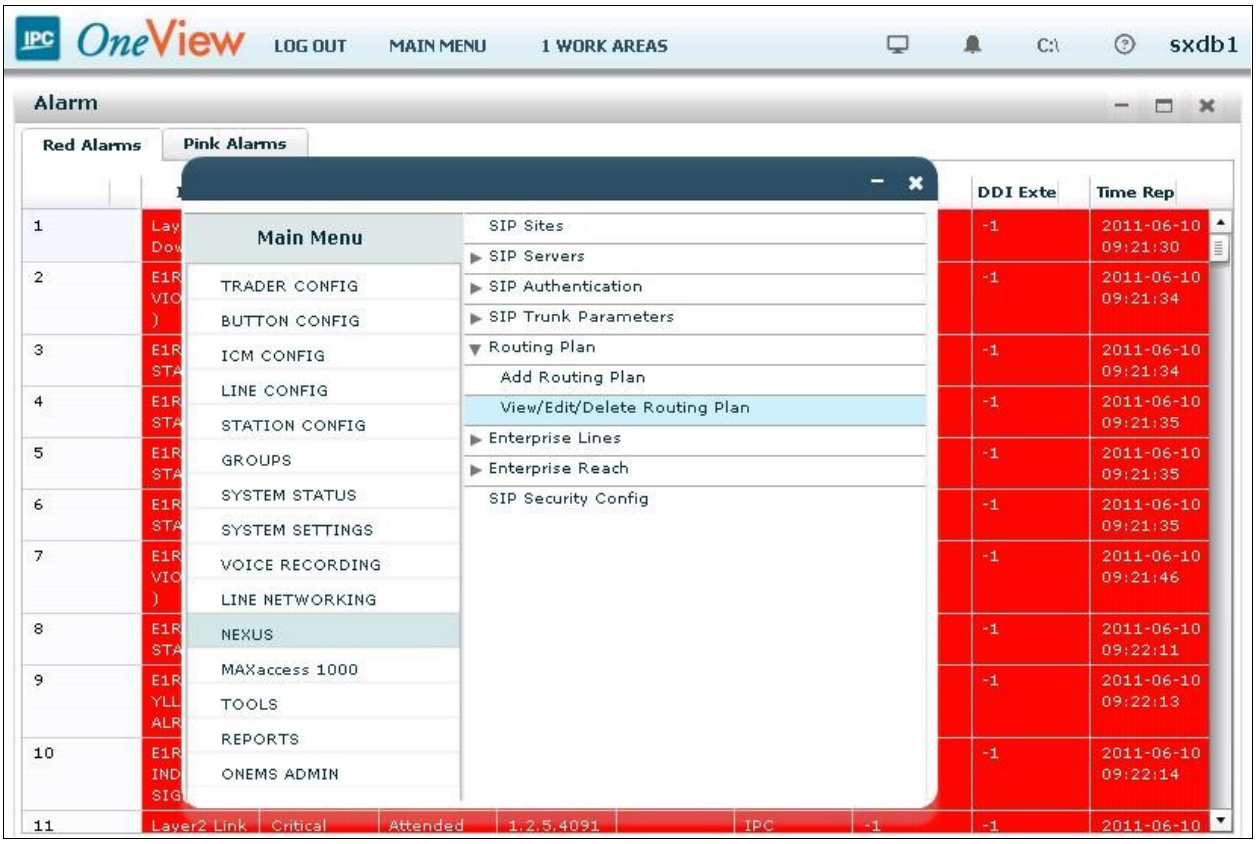



The **Edit SIP Config** screen is displayed. For **DDI Group ID/ DDI Group Name**, select the relevant SIP trunk card number from the drop-down list, in this case "5". Click **Submit**.

The **Edit SIP Config** screen is updated with the located **DDI Group ID** entry. Double click on the **Outbound URL** field corresponding to the located entry, and enter the applicable domain name for the network configuration, in this case "ipc.com". IPC will use this SIP domain in the SIP "From" and "To" headers.


## 6.3. Administer Routing Plan

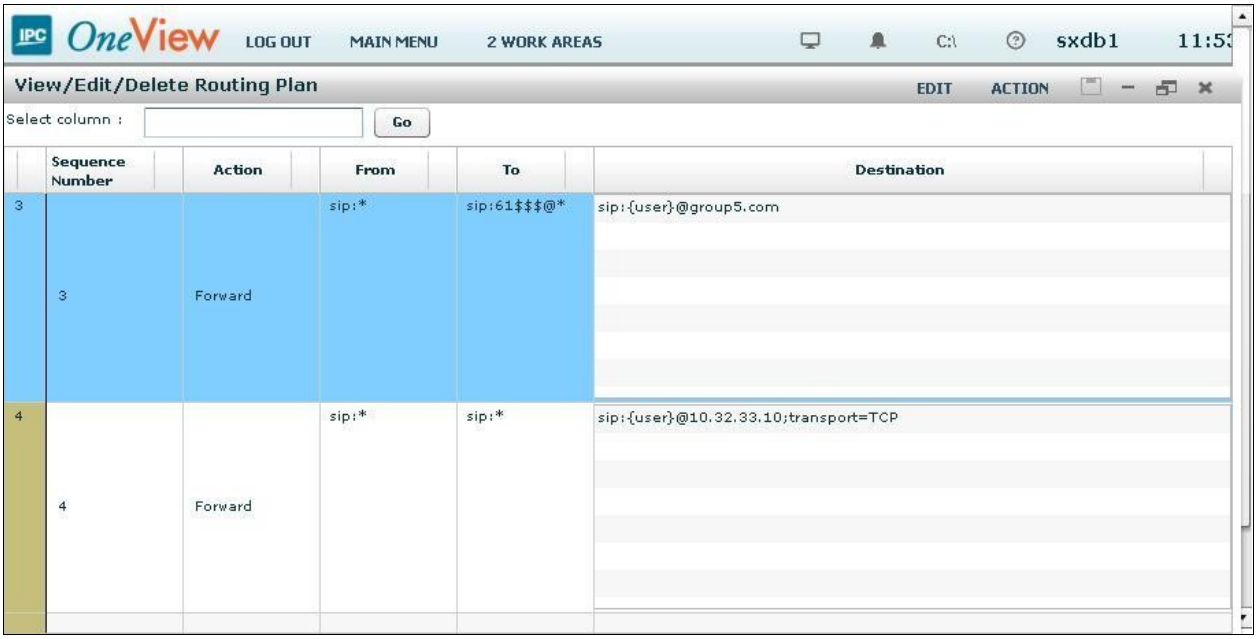
Select **MAIN MENU** from the top menu to display the **Main Menu** screen. Select **NEXUS > Routing Plan > View/Edit/Delete Routing Plan**, as shown below. Click **Submit** in the subsequent screen (not shown), to search for all existing routing plans.



The **View/Edit/Delete Routing Plan** screen is displayed. Follow [2] to add two routing entries shown below.

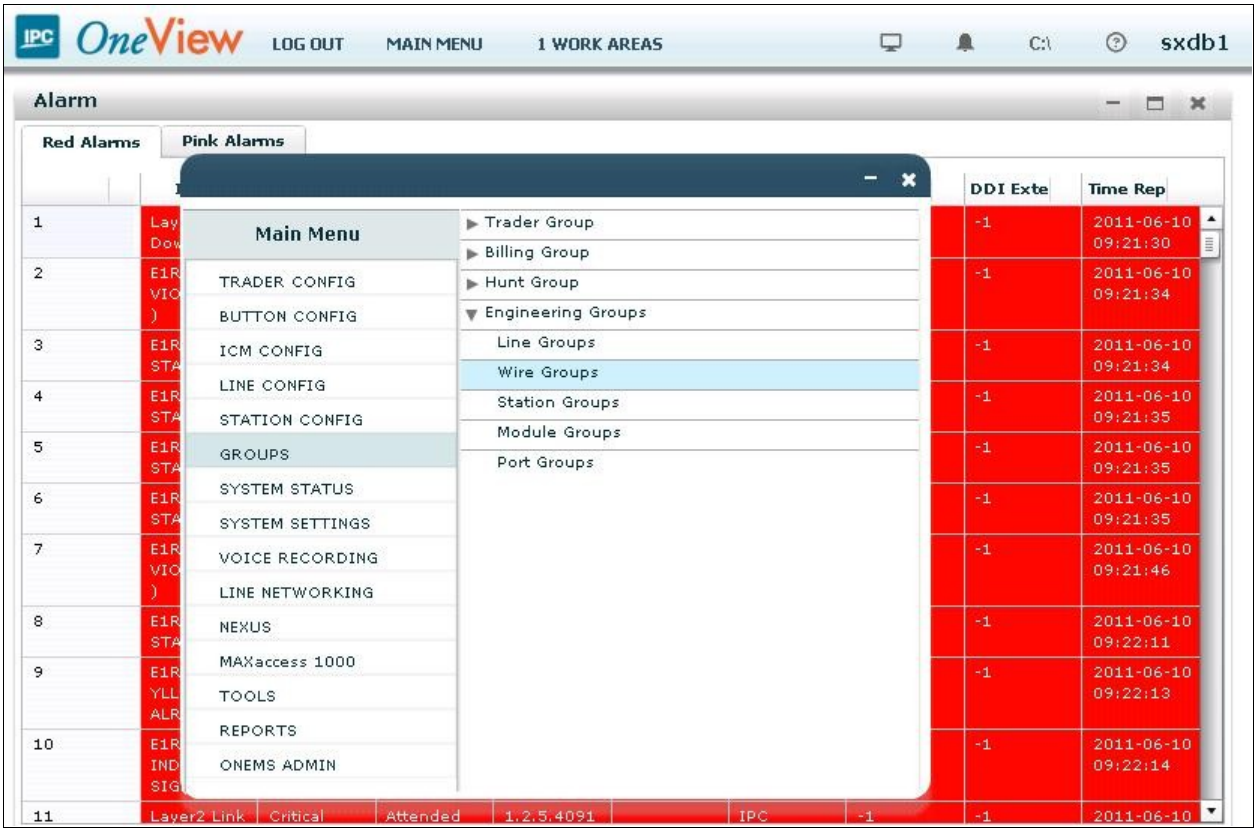
The entry with **Sequence Number 3** was used for routing of inbound calls to IPC. Note that the **Destination** URL contains the internal default value for the SIP trunk card, in this case "group5.com".

The entry with **Sequence Number 4** was used for routing of outbound calls to Avaya IP Office. Note the **Destination** URL includes the IP address of Avaya IP Office, and the transport method from **Section 5.4**.

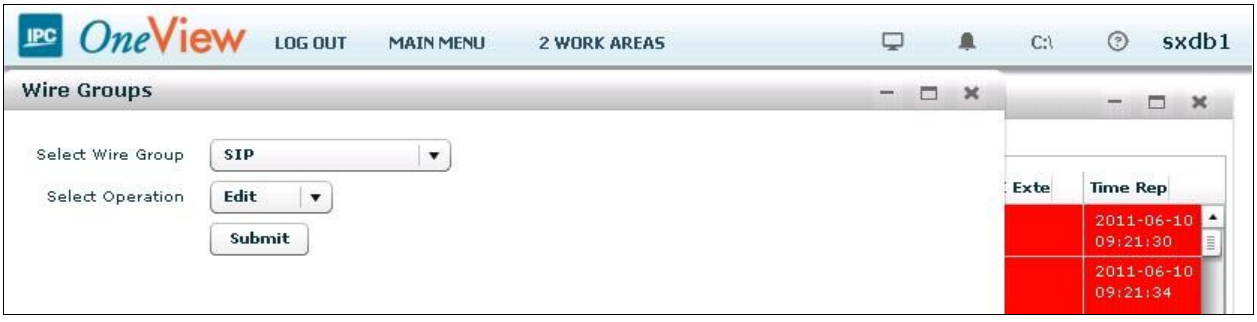


## 6.4. Administer Wire Groups

Select **MAIN MENU** from the top menu to display the **Main Menu** screen. Select **GROUPS > Engineering Groups > Wire Groups**, as shown below.

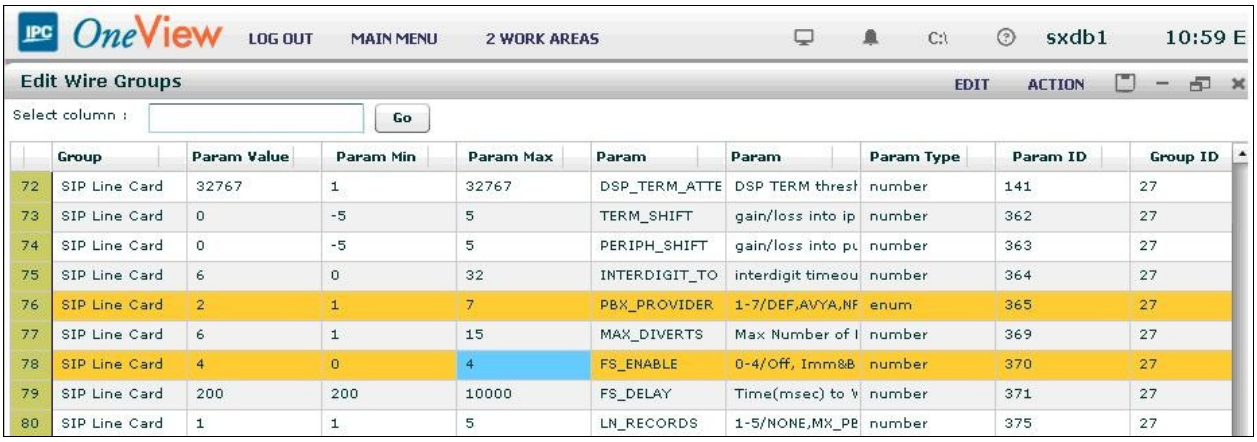



The **Wire Groups** screen is displayed next. Select "SIP" from the **Select Wire Group** drop-down list, and "Edit" from the **Select Operation** drop-down list, as shown below.



The **Edit Wire Groups** screen is displayed. Scroll down the screen as necessary to locate the entry with **Param ID** of "365". Double click on the corresponding **Param Value** field, and enter "2" to denote Avaya as the PBX provider.

Locate the entry with **Param ID** of "370". Double click on the corresponding **Param Value** field, and enter "4" to enable Forward Switching.



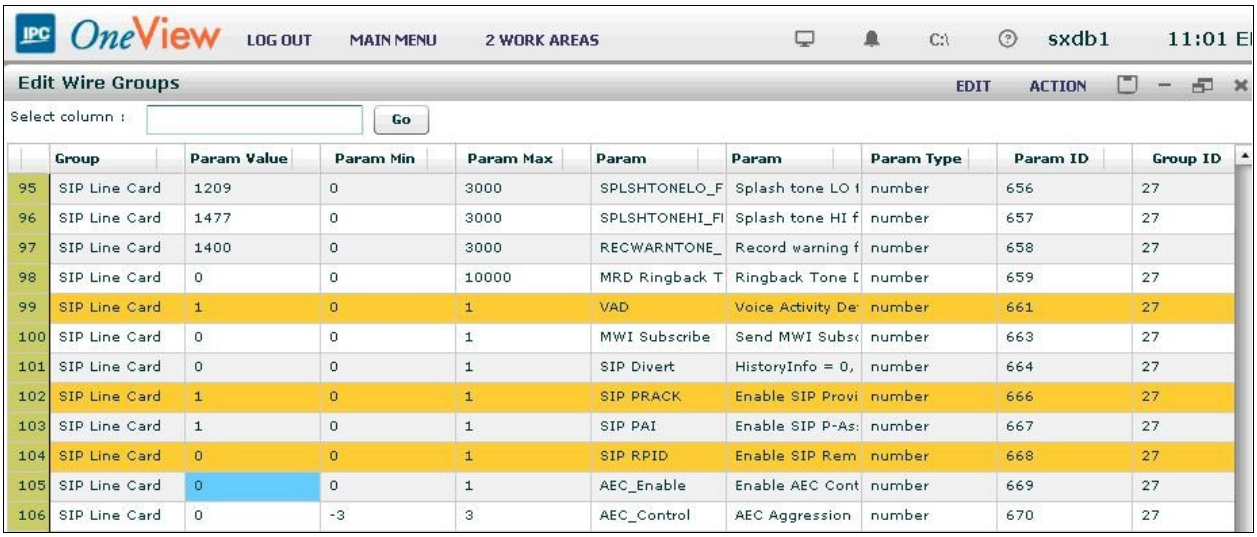

| | Group | Param Value | Param Min | Param Max | Param | Param | Param Type | Param ID | Group ID |
|---|---|---|---|---|---|---|---|---|---|
| 72 | SIP Line Card | 32767 | 1 | 32767 | DSP_TERM_ATTE | DSP TERM thresh | number | 141 | 27 |
| 73 | SIP Line Card | 0 | -5 | 5 | TERM_SHIFT | gain/loss into ip | number | 362 | 27 |
| 74 | SIP Line Card | 0 | -5 | 5 | PERIPH_SHIFT | gain/loss into pu | number | 363 | 27 |
| 75 | SIP Line Card | 6 | 0 | 32 | INTERDIGIT_TO | interdigit timeou | number | 364 | 27 |
| 76 | SIP Line Card | 2 | 1 | 7 | PBX_PROVIDER | 1-7/DEF,AVYA,NF | enum | 365 | 27 |
| 77 | SIP Line Card | 6 | 1 | 15 | MAX_DIVERTS | Max Number of I | number | 369 | 27 |
| 78 | SIP Line Card | 4 | 0 | 4 | FS_ENABLE | 0-4/Off, Imm&B | number | 370 | 27 |
| 79 | SIP Line Card | 200 | 200 | 10000 | FS_DELAY | Time(msec) to V | number | 371 | 27 |
| 80 | SIP Line Card | 1 | 1 | 5 | LN_RECORDS | 1-5/NONE,MX_PE | number | 375 | 27 |

Scroll down the screen as necessary to locate the entry with **Param ID** of "661". Double click on the corresponding **Param Value** field, and enter "1" to activate detection for G729.

Locate the entry with **Param ID** of "666". Double click on the corresponding **Param Value** field, and enter "1" to enable SIP Provisional Acknowledgement (PRACK).

Locate the entry with **Param ID** of "668". Double click on the corresponding **Param Value** field, and enter "0" to disable SIP Remote Party ID (RPI).

Follow [2] to reboot the SIP trunk card.




| | Group | Param Value | Param Min | Param Max | Param | Param | Param Type | Param ID | Group ID |
|---|---|---|---|---|---|---|---|---|---|
| 95 | SIP Line Card | 1209 | 0 | 3000 | SPLSHTONELO_F | Splash tone LO f | number | 656 | 27 |
| 96 | SIP Line Card | 1477 | 0 | 3000 | SPLSHTONEHI_FI | Splash tone HI f | number | 657 | 27 |
| 97 | SIP Line Card | 1400 | 0 | 3000 | RECWARNTONE_ | Record warning f | number | 658 | 27 |
| 98 | SIP Line Card | 0 | 0 | 10000 | MRD Ringback T | Ringback Tone D | number | 659 | 27 |
| 99 | SIP Line Card | 1 | 0 | 1 | VAD | Voice Activity De | number | 661 | 27 |
| 100 | SIP Line Card | 0 | 0 | 1 | MWI Subscribe | Send MWI Subsc | number | 663 | 27 |
| 101 | SIP Line Card | 0 | 0 | 1 | SIP Divert | HistoryInfo = 0, | number | 664 | 27 |
| 102 | SIP Line Card | 1 | 0 | 1 | SIP PRACK | Enable SIP Provi | number | 666 | 27 |
| 103 | SIP Line Card | 1 | 0 | 1 | SIP PAI | Enable SIP P-As: | number | 667 | 27 |
| 104 | SIP Line Card | 0 | 0 | 1 | SIP RPID | Enable SIP Rem | number | 668 | 27 |
| 105 | SIP Line Card | 0 | 0 | 1 | AEC_Enable | Enable AEC Cont | number | 669 | 27 |
| 106 | SIP Line Card | 0 | -3 | 3 | AEC_Control | AEC Aggression | number | 670 | 27 |

## 6.5. 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 Avaya IP Office as a trusted host. Note that 10.32.33.10 is the IP address of Avaya IP Office.

The same command can be used with the "-view" option to make certain Avaya IP Office is displayed as a trusted host.

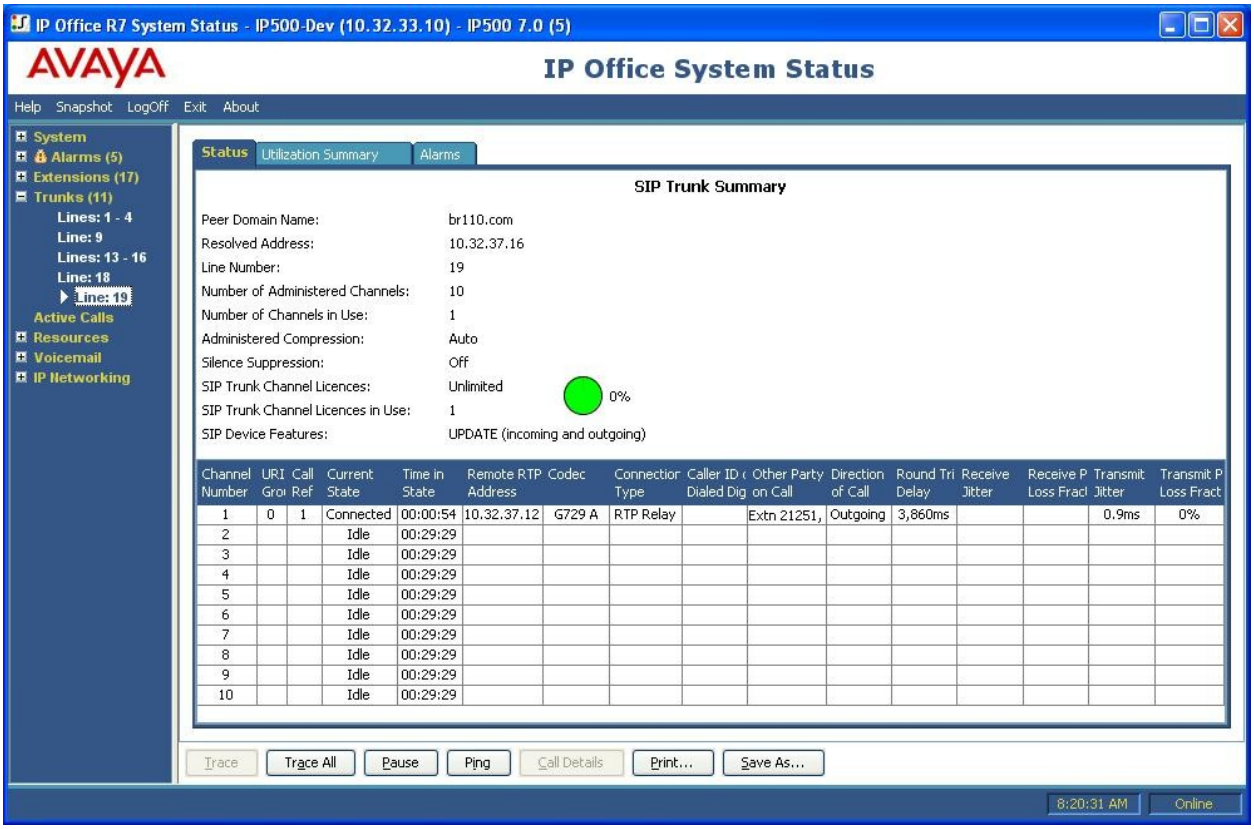
```
[root@esshost ~]# cd /usr/local/SipProxy/
[root@esshost SipProxy]# ./trusted_hosts.pl -add=10.32.33.10

[root@esshost SipProxy]# ./trusted_hosts.pl -view
ip_address      last_modified
10.32.33.10     2011-06-03 13:45:24
```

# 7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya IP Office and IPC System Interconnect. Establish a call between Avaya IP Office and IPC System Interconnect.

From the **Avaya IP Office R7 Manager** screen shown in **Section 5.1**, select **File > Advanced > System Status** to launch the System Status application, and log in using the appropriate credentials. The **IP Office System Status** screen is displayed. Expand **Trunks** in the left pane and select the SIP line from **Section 5.4**, in this case "19".

Verify that the **SIP Trunk Summary** screen shows an active channel with **Current State** of "Connected". Also verify that the **Remote RTP Address** contains the IP address of the IPC Alliance MX, and that **the Other Party on Call** contains the local IPO user.



# 8. Conclusion

These Application Notes describe the configuration steps required for IPC System Interconnect to successfully interoperate with Avaya IP Office. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

# 9. Additional References

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

1. *IP Office 7.0 Documentation CD*, March 2011, available at http://support.avaya.com.

2. *Nexus Suite 2.0 SP1 Patch11 or Higher Deployment Guide*, Part Number B02200161, Revision Number 01, 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.