



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for IPC Unigy with Avaya IP Office using SIP Trunks – Issue 1.0**

### **Abstract**

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

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

The embedded IP Office Voicemail 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 Unigy to interoperate with Avaya IP Office.

IPC Unigy is a trading communication solution. In the compliance testing, IPC Unigy 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 Unigy 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 IPC Unigy.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

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 Unigy to recover from adverse conditions, such as disconnecting/reconnecting the LAN connection to IPC Unigy.

## 2.2. Test Results

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

- Blind Transfer – Avaya H.323 calls IPC turret, and IPC transfers to PSTN failed. However, consult transfer worked for the same scenario. An extension (H323) calls an IPC turret, and the IPC turret answers the call. The turret transfers the call to PSTN. The phone on the PSTN rings. As the phone is picked up, the phone goes on-hook. The Avaya H.323 phone keeps ringing.
- Conference – IPC does not support initiating conference. The work around is performing conference at the IP Office side.

## 2.3. Support

Technical support on IPC Unigy can be obtained through the following:

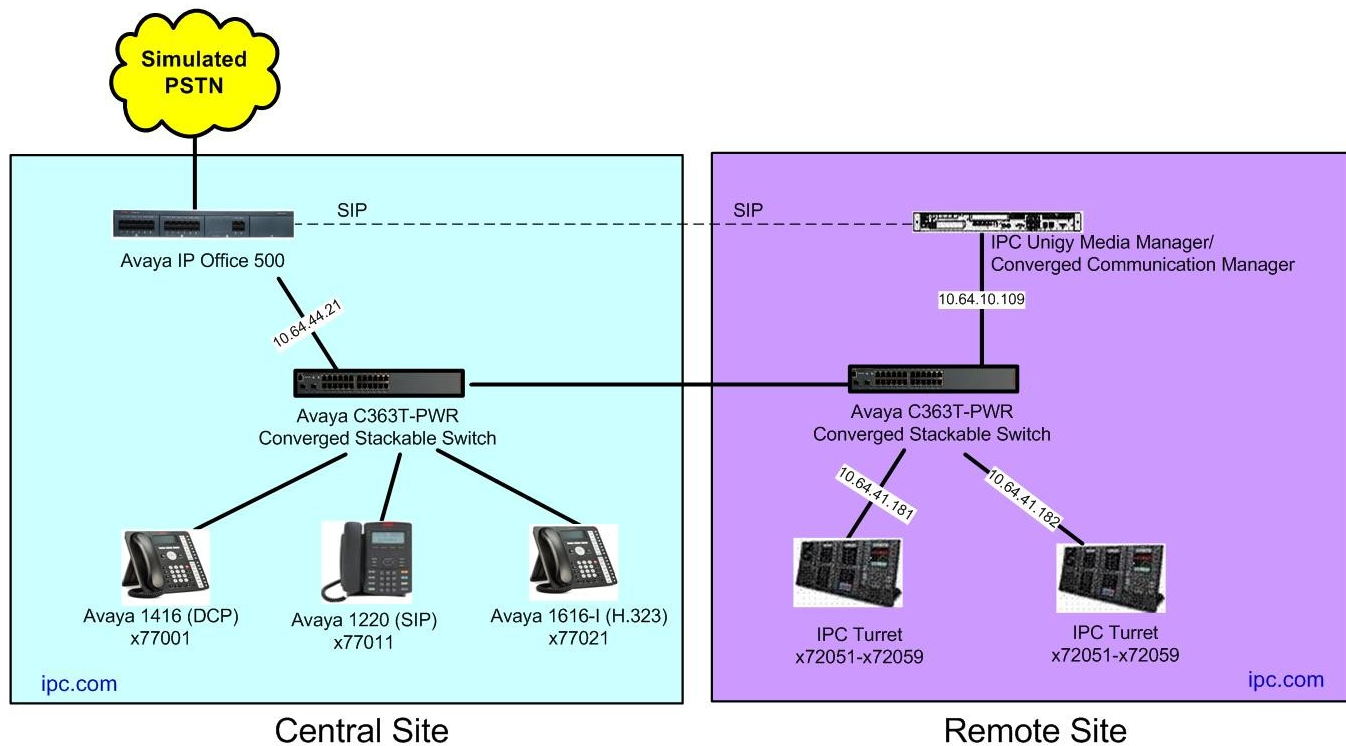
- **Phone:** (800) NEEDIPC, (203) 339-7800
- **Email:** [systems.support@ipc.com](mailto:systems.support@ipc.com)

## 3. Reference Configuration

As shown in **Figure 1**, IPC Unigy at the Remote Site consists of the Media Manager, Converged Communication Manager, and Turrets. The Media Manager and Converged Communication Manager are typically deployed on separate servers. In the compliance testing, the same server hosted the Media Manager and Converged Communication Manager.

The embedded IP Office Voicemail 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 (7700x, 7701x and 7702x), and IPC turret users at the Remote site (7205x).



**Figure 1: Test Configuration of IPC Unigy**

## 4. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office 500 V2	8.0 (16)
Avaya IP Office Manager	10.0 (16)
Avaya1616-I (H.323)	1.3.0
Avaya 1416 Digital Telephone	-
Avaya 1220 IP Deskphone (SIP)	04.03.09.00
IPC Unigy	01.00.00.04.0009
Turrets	01.00.00.04.0009

## 5. Configure Avaya IP Office

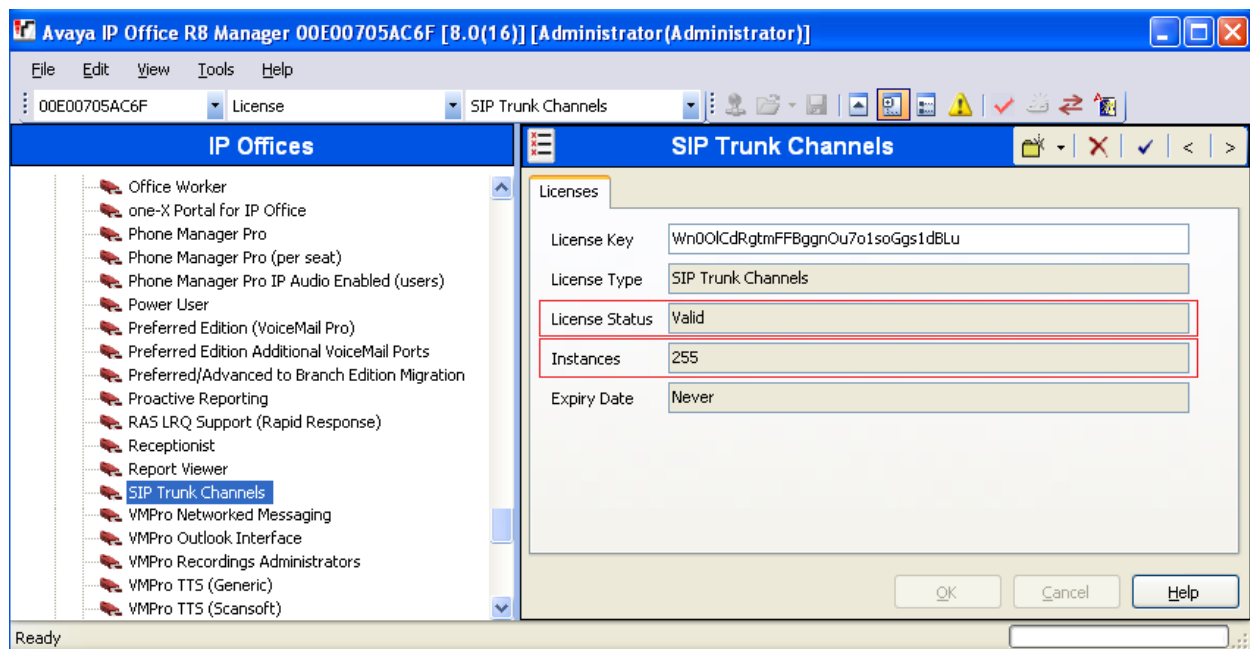
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- 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 R8 Manager** screen is displayed. From the configuration tree in the left pane, select **License → SIP Trunk Channels** to display the **SIP Trunk Channels** screen in the right pane. Verify that the **License 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 screen below 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.

The screenshot shows the IP Office configuration interface. On the left is a tree view of the system configuration. The right pane is titled '00E00705AC6F' and contains several tabs: VCM, CCR, Codecs, System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, and Twinning. The 'LAN1' tab is selected, and within it, the 'LAN Settings' sub-tab is active. The 'LAN Settings' sub-tab contains the following fields:

- IP Address: 10 . 64 . 44 . 21
- IP Mask: 255 . 255 . 255 . 0
- Primary Trans. IP Address: 0 . 0 . 0 . 0
- RIP Mode: None
- Enable NAT: ☐
- Number Of DHCP IP Addresses: 200
- DHCP Mode: ☐ Server ☐ Client ☐ Dialin ☒ Disabled

An 'Advanced' button is located at the bottom right of the LAN Settings sub-tab.

## 5.3. Enable SIP Trunks

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

The screenshot shows the IP Office configuration interface. On the left is a tree view of the system configuration. The right pane is titled '00E00705AC6F' and contains several tabs: VCM, CCR, Codecs, System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, and Twinning. The 'LAN1' tab is selected, and within it, the 'VoIP' sub-tab is active. The 'VoIP' sub-tab contains the following fields:

- ☒ H.323 Gatekeeper Enable
- ☒ SIP Trunks Enable
- ☒ SIP Registrar Enable
- ☐ H.323 Auto-create Extn
- ☐ H.323 Auto-create User
- ☐ H.323 Remote Extn Enable
- ☒ Enable RTPC Monitoring On Port 5005
- RTP Port Number Range:
  - Port Range (Minimum): 49152
  - Port Range (Maximum): 53246

## 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 “ipc.com”. Uncheck **REFER Support**, as shown below. Retain the default values in the remaining fields.

**IP Offices**

- BOOTP (2)
- Operator (3)
- 00E00705AC6F
  - System (1)
  - Line (13)
    - 1
    - 2
    - 3
    - 4
    - 5
    - 17
    - 21
    - 22
    - 24
    - 25
    - 26
    - 27
    - 28
  - Control Unit (3)
  - Extension (22)
  - User (24)
  - HuntGroup (2)

**SIP Line - Line 28**

**SIP Line** | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials

Line Number: 28

ITSP Domain Name: ipc.com

In Service: ☒

Use Tel URI: ☐

Check OOS: ☒

Call Routing Method: Request URI

Originator number for forwarded and twinning calls:

Name Priority: System Default

Association Method: By Source IP address

☐ REFER Support

Incoming: Auto

Outgoing: Auto

Select the **Transport** tab in the right pane. For **ITSP Proxy Address**, enter the IP address of IPC Unigy. For **Layer 4 Protocol**, select “UDP”. Retain the default values for the remaining fields.

**IP Offices**

- BOOTP (2)
- Operator (3)
- 00E00705AC6F
  - System (1)
  - Line (13)
    - 1
    - 2
    - 3
    - 4
    - 5
    - 17
    - 21
    - 22
    - 24
    - 25
    - 26
    - 27
    - 28
  - Control Unit (3)
  - Extension (22)
  - User (24)
  - HuntGroup (2)

**SIP Line - Line 28**

**SIP Line** | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials

ITSP Proxy Address: 10.64.10.109

**Network Configuration**

Layer 4 Protocol: UDP

Send Port: 5060

Use Network Topology Info: None

Listen Port: 5060

Explicit DNS Server(s): 0 . 0 . 0 . 0 0 . 0 . 0 . 0

Calls Route via Registrar: ☒

Separate Registrar:

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.

**SIP Line - Line 28**

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls

**Edit Channel**

Via: <None>

Local URI: \*

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 28

Outgoing Group: 28

Max Calls per Channel: 10

OK Cancel

The following screen shows the SIP URI page used during the compliance test.

**SIP Line - Line 28**

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	28 28	<...*	*	*	*	N...	0: <Non...	10
2	28 28	<...*	*	*	*	N...	0: <Non...	10

Add... Remove Edit...



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

The screenshot shows the 'SIP Line - Line 28' configuration window. The 'VoIP' tab is active. In the 'Selected' codec list, several codecs are listed. The 'Re-invite Supported' checkbox is checked and highlighted with a red box. Other settings include 'Codec Selection' set to 'System Default', 'Fax Transport Support' set to 'None', 'Call Initiation Timeout (s)' set to '4', and 'DTMF Support' set to 'RFC2833'.

## 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 “28”.

The screenshot shows the 'Incoming Call Route' configuration window. The 'Line Group ID' is set to '28' and is highlighted with a red box. The 'Destinations' tab is visible, showing a table with columns for 'TimeProfile', 'Destination', and 'Fallback Extension'. The 'Default Value' row is highlighted.

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

The screenshot shows the 'Incoming Call Route' configuration window, 'Destinations' tab. The 'Destination' field is set to '.' and is highlighted with a red box. The 'Fallback Extension' field is also visible.

## 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 7205x, and the calls are routed over the SIP trunk to IPC Unigy.

For **Code**, enter “7205x”. For **Telephone Number**, enter the value shown below where “.” is to denote any calls that starts with 7205 will be sent using Line group (trunk) 28. For **Line Group ID**, enter the outgoing group number from **Section 5.4**.

The screenshot shows the '7205x: Dial' configuration window. The left pane lists 'IP Offices' with a tree structure including various extension ranges. The main pane is titled '7205x: Dial' and contains the following fields:

- Code:** 7205x
- Feature:** Dial (dropdown menu)
- Telephone Number:** .
- Line Group ID:** 28 (dropdown menu)
- Locale:** (dropdown menu)
- Force Account Code:** ☐

## 5.7. Administer Users

From the configuration tree in the left pane, select a user from **Section 3** that will be placing and receiving calls via the SIP trunks with IPC. In this case, the user is “77011”. 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 77011, 77022, and 77001 were configured.

The screenshot shows the 'Ext211: 77011' configuration window. The left pane lists 'IP Offices' with a tree structure including various extension ranges. The main pane is titled 'Ext211: 77011' and contains the following fields:

- SIP Name:** 77011
- SIP Display Name (Alias):** Ext211
- Contact:** 77011
- Anonymous:** ☐

## 6. Configure IPC Media Manager

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

- Launch Unigy Management System
- Administer SIP trunks
- Administer trunk group
- Administer route lists
- Administer dial patterns
- Administer route plans

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

### 6.1. Launch Unigy Management System

Access the Unigy Management System web interface by using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the Media Manager. Log in using the appropriate credentials.

The screen below is displayed. Enter the appropriate credentials. Check **I agree with the Terms of Use**, and click **Login**.

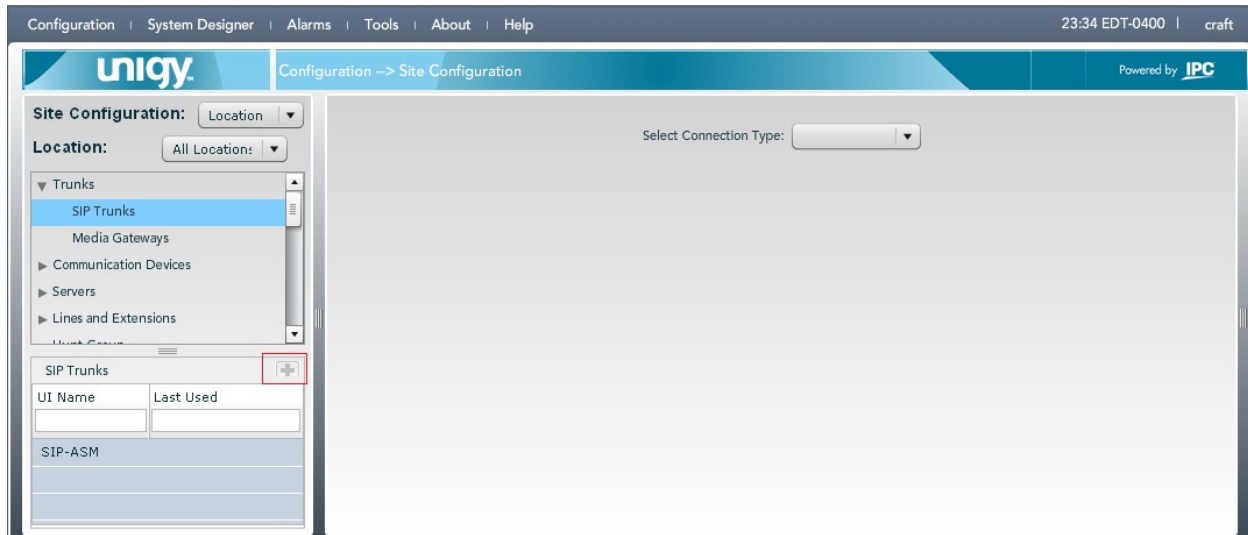
In the subsequent screen (not shown), click **Continue**.



The screenshot shows the login interface for the IPC Unigy Management System. It features the IPC logo on the left. To the right of the logo are two input fields: 'User Name:' and 'Password:'. Below these fields is a checkbox labeled 'I agree with the' followed by a link to 'Terms of Use'. A 'Login' button is positioned to the right of the checkbox. At the bottom of the form, the text reads: 'IPC Unigy™ Management System', 'Unigy™ Version 01.00.00.04.0009', and '© Copyright 2011 IPC Systems, Inc.'

## 6.2. Administer SIP Trunks

Select **Site Configuration** under the **Configuration** menu at the top. Navigate to **Trunks** → **SIP Trunks** in the left pane. Click the **Add** icon in the lower left pane to add a new SIP trunk. The screen below is displayed. Select “Dial Tone” from the **Select Connection Type** drop-down list.



Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Trunk Name** A descriptive name
- **Destination Address** IP address of IP Office.
- **Destination Port** "5060"
- **Zone** An available zone, in this case "Default Zone 1".
- **Channel** The number of SIP trunk group members in **Section 5.4**.
- **Reason Protocol** "SIP"
- **PBX Provider** "Avaya"
- **Connected Party Update** "UPDATE"

Click the **Save** button.

The screenshot displays the UniQy configuration interface. The top navigation bar includes links for Configuration, System Designer, Alarms, Tools, About, and Help. The current page is titled "Configuration -> Site Configuration". On the left, a sidebar shows a tree view of configuration options: Trunks, SIP Trunks (selected), Media Gateways, Communication Devices, Servers, Lines and Extensions, Hunt Group, and Routing. Below this, a table lists SIP Trunks with columns for UI Name and Last Used. The main area is titled "Trunk: DialTone" and contains a "Trunk Configuration" form. The form fields are as follows:

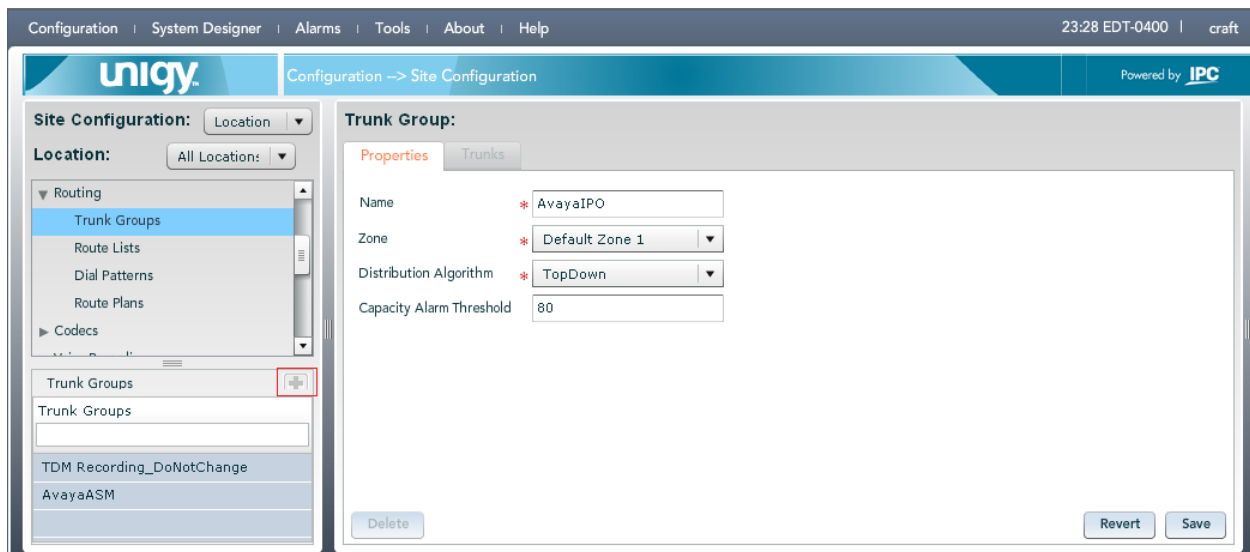
Field	Value
Trunk Name	SIP-IPO
Number of Trunks	1
Connection Type	Dial Tone
Destination Address	10.64.44.21
Destination Port	5060
Media Manager Profile	Safe
Zone	Default Zone 1
Channels	10
Reason Protocol	SIP
PBX Provider	None
Connected Party Update	UPDATE

At the bottom right of the form are buttons for Delete, Revert, and Save.

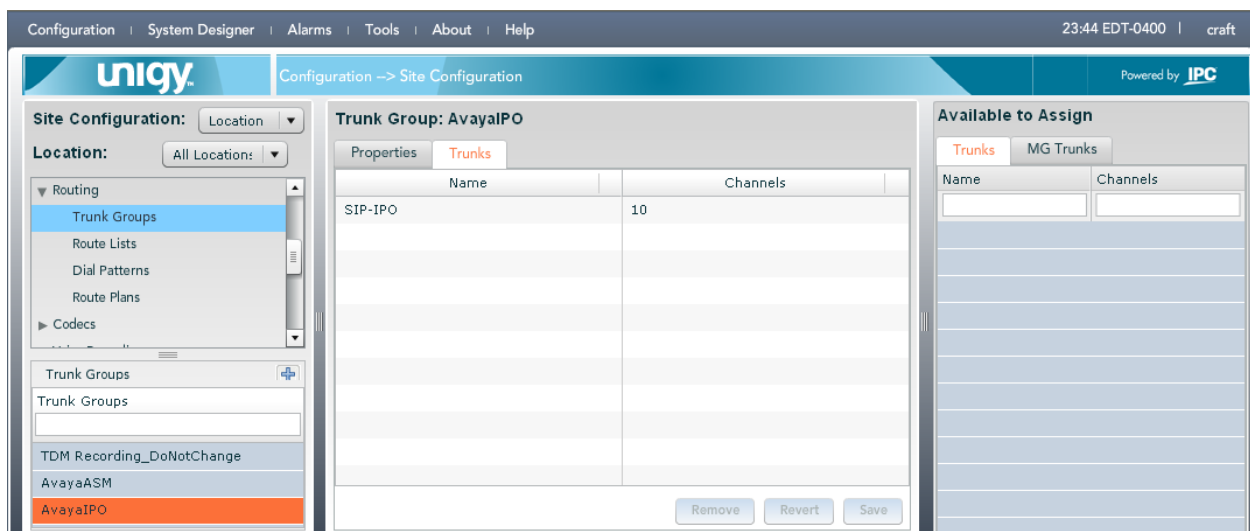
### 6.3. Administer Trunk Groups

Select **Routing** → **Trunk Groups** in the left pane, and click the **Add** icon in the lower left pane to add a new trunk group.

The **Trunk Group** screen is displayed in the right pane. In the **Properties** tab, enter a descriptive **Name**, and click **Save**. Select the **Trunks** tab in the right pane.



The screen is updated with three panes. In the rightmost pane, select the **Trunks** tab to display a list of trunks. Select the SIP trunk from **Section 6.2** in the rightmost pane and drag to the middle pane as shown below. Click **Save**.



## 6.4. Administer Route Lists

Select **Routing** → **Route Lists** in the left pane, and click the **Add** icon in the lower left pane to add a new route list.

The **Route List** screen is displayed in the middle pane. For **Route List**, enter a descriptive name. In the right pane, select the trunk group from **Section 6.3** and drag into the **Assigned Trunk Groups on Route List** sub-section in the middle pane, as shown below. Click **Save**.

The screenshot displays the UniQy configuration interface for a Route List. The top navigation bar includes 'Configuration', 'System Designer', 'Alarms', 'Tools', 'About', and 'Help'. The main header shows 'Configuration -> Site Configuration' and 'Powered by IPC'. The left pane, titled 'Site Configuration', shows a tree view with 'Routing' expanded and 'Route Lists' selected. Below this, a 'Route Lists' section shows a list of route lists with 'Route2ASM' currently selected. The middle pane, titled 'Route List : Route List', contains a form with 'Route List' set to 'Route2IPO' and 'Description' empty. Below the form is a section titled 'Assigned Trunk Groups on Route List. You can remove or add Trunk Groups' which contains a list with 'AvayaIPO'. At the bottom of the middle pane are 'Revert', 'Delete', and 'Save' buttons. The right pane, titled 'Available to Assign', shows a list of trunk groups with 'AvayaIPO' selected. The top right of the interface shows the time '23:46 EDT-0400' and the user 'craft'.

## 6.5. Administer Dial Patterns

Select **Routing → Dial Patterns** in the left pane, to display the **Dial Patterns** screen in the right pane. Click **Add New** in the upper right pane.

In the **Dial pattern Details** sub-section in the lower right pane, enter the desired **Name** and **Description**. For **Pattern String**, enter “\*”, meaning any call will be sent to IP Office. For **Call Classification**, select “External”.

Click **Save**.

The screenshot shows the UniQy Configuration interface. The left pane displays the 'Site Configuration' tree with 'Dial Patterns' selected. The right pane shows the 'Dial Patterns' section with a table of dial patterns and a 'Dial pattern Details' form. The 'Add New' button is highlighted with a red box.

Name	Pattern String	Outbound CLI	Call Classification	Prefix Digits	Description

**Dial pattern Details**

**Properties**

Name \* all

Description \* Call to IPO

Pattern String \* \*

Outbound CLI

Call Classification \* External

Prefix Digits

Revert Save

In the compliance the following dial pattern was created.

The screenshot shows the UniQy Configuration interface. The left pane displays the 'Site Configuration' tree with 'Dial Patterns' selected. The right pane shows the 'Dial Patterns' section with a table of dial patterns. One entry is visible in the table.

Name	Pattern String	Outbound CLI	Call Classification	Prefix Digits	Description
all	*		External		*



## 6.6. Administer Route Plans

Select **Routing** → **Route Plans** in the left pane, and click **Add New** (not shown) in the right pane to create a new route plan.

The screen is updated with three panes, as shown below. In the **Route Plan** middle pane, enter a descriptive **UI Name** and optional **Description**. For **Calling Party**, enter “\*” to denote any calling party from Unigy. For **Called Party**, select ”all”. Select “Forward” for **Action**, and click **Save**.

The screenshot shows the Unigy configuration interface. The left pane is titled 'Site Configuration' and shows a tree view with 'Route Plans' selected. The middle pane is titled 'Route Plan' and contains a 'Create New Route Plan' form. The form has the following fields:

- UI Name: \* Route to IPO
- Description: (empty)
- Calling Party: \*
- Called Party: \* all
- Action: \* Forward
- Route List: (empty table)

Buttons at the bottom of the form include 'Back', 'Revert', and 'Save'. The right pane is titled 'Available to Assign' and shows a list of route lists with 'Route2IPO' selected.

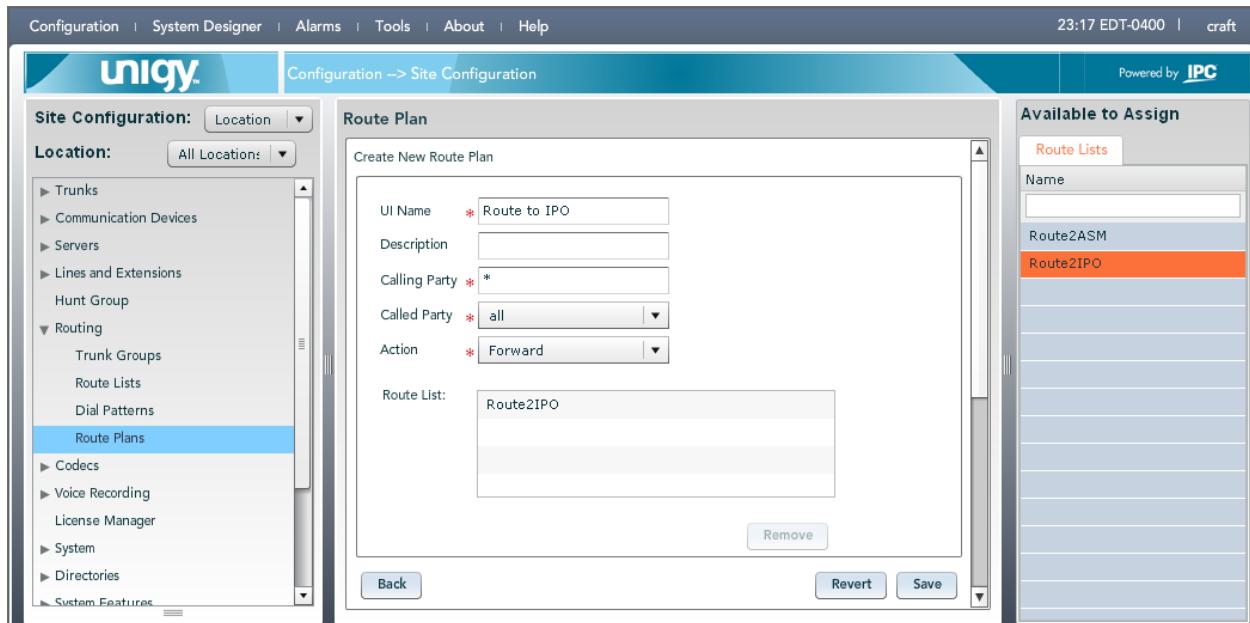
The screen is updated with the newly created route plan. Select the route plan, and click **Edit** toward the bottom of the screen (not shown).

The screenshot shows the Unigy configuration interface with the 'List of Route Plans' table. The table has the following columns: UI Name, Calling Party, Called Party, and Action. The data row shows:

UI Name	Calling Party	Called Party	Action
Route to IPO	*	all	FORWARD

Buttons at the bottom of the table include 'Delete', 'Add New', 'Revert', and 'Save Sequence Change'.

The screen is updated with three panes again, as shown below. In the right pane, select the route list from **Section 6.4** and drag into the **Route List** sub-section in the middle pane, as shown below. Click **Save**.



## 7. Verification Steps

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

From the PC that installed **Avaya IP Office R7 Manager**, navigate to **All Programs → IP Office → 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 “28”.

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 turret, and that the **Other Party on Call** contains the local IPO user.

The screenshot displays the Avaya IP Office System Status application. The left sidebar shows a tree view with 'System' expanded, containing 'Alarms (35)', 'Extensions (12)', 'Trunks (13)', and 'Active Calls'. Under 'Trunks', 'Line: 28' is selected. The main window shows the 'SIP Trunk Summary' for Line 28. The summary includes fields for Peer Domain Name (ipc.com), Resolved Address (10.64.10.109), Line Number (28), Number of Administered Channels (20), Number of Channels in Use (1), Administered Compression (G729 A, G711 Mu, G711 A, G7231), Silence Suppression (Off), SIP Trunk Channel Licenses (Unlimited), and SIP Trunk Channel Licenses in Use (1). A green circle indicates 0% utilization. Below the summary is a table of channel states.

Channel Number	URI G...	Call Ref	Current State	Time in State	Remote Media Addr...	Codec	Connection Type	Caller ID or Dial...	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Packet L...	Transmit Jitter	Transmit Packet L...
1	0	2615	Connected	00:00:33	10.64.41.181	G729 A	RTP Relay		Extn 77011, Extn21	Outgoing					
2			Idle	39 days ...											
3			Idle	41 days ...											
4			Idle	41 days ...											
5			Idle	42 days ...											
6			Idle	42 days ...											
7			Idle	42 days ...											
8			Idle	42 days ...											
9			Idle	42 days ...											
10			Idle	42 days ...											
11			Idle	42 days ...											
12			Idle	42 days ...											
13			Idle	42 days ...											
14			Idle	42 days ...											
15			Idle	42 days ...											
16			Idle	42 days ...											
17			Idle	42 days ...											
18			Idle	42 days ...											
19			Idle	42 days ...											
20			Idle	42 days ...											

## 8. Conclusion

These Application Notes describe the configuration steps required for IPC Unigy to successfully interoperate with Avaya IP Office. All feature and serviceability test cases were completed. 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 8.0 IP Office Installation*, Document Number 15-601042, Issue 25b, March 08, 2012
2. *IP Office Release 8.0 Manager 10.0*, Document Number 15-601011, Issue 28h, March 28 2012
3. *] IP Office System Status Application*, Issue 06b, November 12, 2011 Document Number 15-601758
4. *IP Office System Monitor*, Document Number 15-601019, Issue 02b
5. *Unigy 1.0 System Configuration*, Part Number B02200187, Release 01, upon request to IPC Support.

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