



Avaya Solution & Interoperability Test Lab

Application Notes for Gamma Telecom IP DirectConnect and Avaya IP Office 6.1 – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the Gamma Telecoms' IP DirectConnect and Avaya IP Office. The Gamma IP DirectConnect provides PSTN access via a SIP trunk connected to the Gamma Voice Over Internet Protocol (VoIP) network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise. Gamma Telecom is a member of the Avaya DevConnect Service Provider program.

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 procedures for configuring Session Initiation Protocol (SIP) trunking between the Gamma Telecom IP DirectConnect SIP Trunking service and Avaya IP Office. IP DirectConnect service provides PSTN access via a SIP trunk connected to the Gamma Telecom Voice Over Internet Protocol (VoIP) network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using IP Office to connect to the Gamma Telecom DirectConnect SIP Trunking service. The configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

2.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to the Gamma Telecoms DirectConnect SIP Trunking service. To verify SIP trunking interoperability the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types.
- Phone types included H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider.
- Outgoing PSTN calls from various phone types.
- Phone types included H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider.
- Inbound and outbound PSTN calls to/from Phone Manager Lite clients.
- Various call types including: local, long distance, international and directory assistance (118118)
- Codecs G.729A and G.711A
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning
- T.38 fax

2.2. Test Results

Gamma Telecom SIP Direct Trunking service successfully passed interoperability testing. It should be noted that the following was not tested:

- Toll free inbound and outbound calls were not tested.

2.3. Support

For technical support on Gamma Telecom products please contact the Gamma Telecom support team at:

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3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Gamma Telecom IP DirectConnect service. Located at the enterprise site is an Avaya IP Office 500. Endpoints include an Avaya 9600 Series IP Telephones (with H.323 firmware), an Avaya 5600 Series IP Telephone (with H.323 firmware), an Avaya 5420 Digital Telephone, an Avaya 98390 Analog Telephone and fax machine. The site also has a Windows XP PC running Avaya IP Office Manager to configure the IP Office. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with arbitrary numbers that bear no relevance to the test configuration.

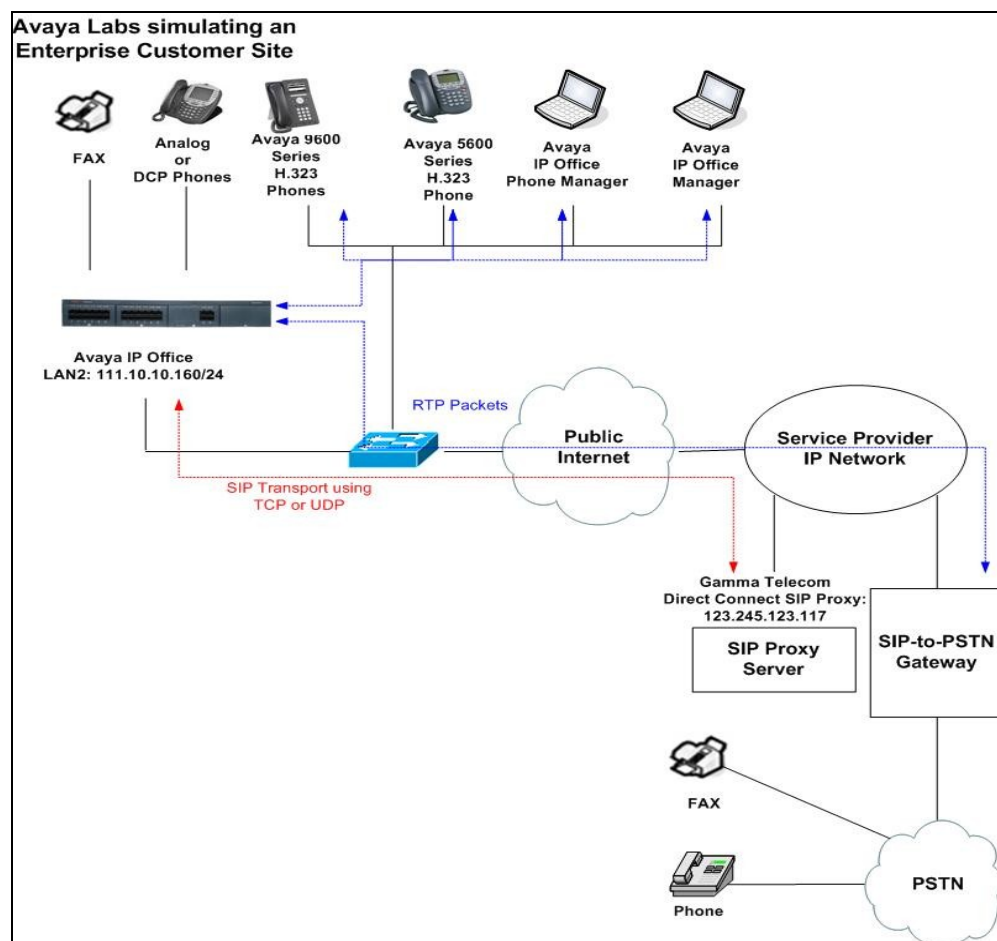


Figure 1: Gamma Telecom Sample Configuration

IP Office was configured to connect to a static IP address at the service provider instead of performing a SIP registration to Gamma Telecom. Gamma Telecom did not challenge messages sent from IP Office and vice versa. For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Gamma Telecom. The short code of 9 is stripped off by IP Office but the remaining N digits were sent unaltered to Gamma Telecom.

In an actual customer configuration, the enterprise site may also include additional network components between the Service Provider and IP Office such as a session border controller or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and IP Office must be allowed to pass through these devices. Gamma Telecom sends SIP signaling from one IP address. However, RTP traffic may originate from a different IP address and ports which may vary from customer to customer. Customers will need to work with Gamma Telecom to determine the proper IP addresses and ports that require access to their network.

4. Equipment and Software Validated

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

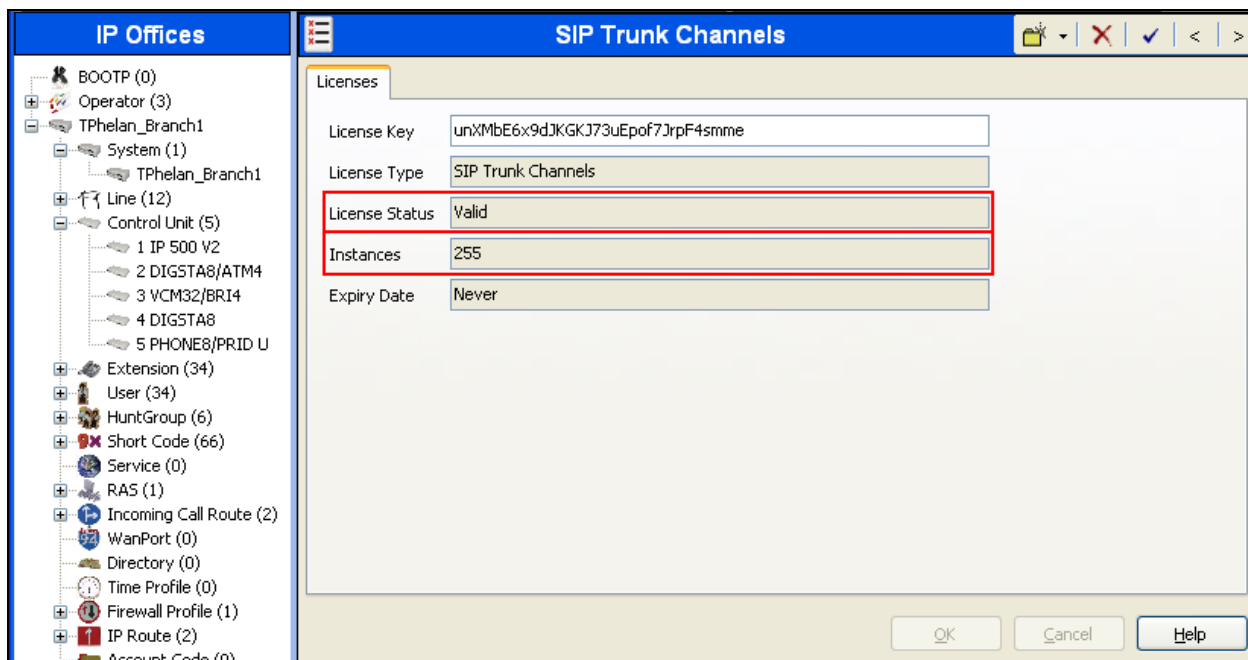
Equipment	Software
Avaya IP Office 500 V2	6.1(5)
Avaya IP Office Manager	8.1(5)
Avaya IP Office Phone Manager	4.2(37)
Avaya 9630 Phone (H.323)	3.11
Avaya 5621 Phone (H.323)	2.9.1
Avaya 5420 Digital Phone	NA
Avaya 98390 Analog Phone	NA

5. Configure Avaya IP Office

This section describes the IP Office configuration to support connectivity to the Gamma Telecom IP DirectConnect SIP Trunking service. IP Office is configured through the IP Office Manager PC application. From a PC running the IP Office Manager application, select **Start→Programs→IP Office→Manager** to launch the application. Navigate to **File→Open Configuration**, select the proper IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one in the next section. All the IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning) is assumed to already be in place.

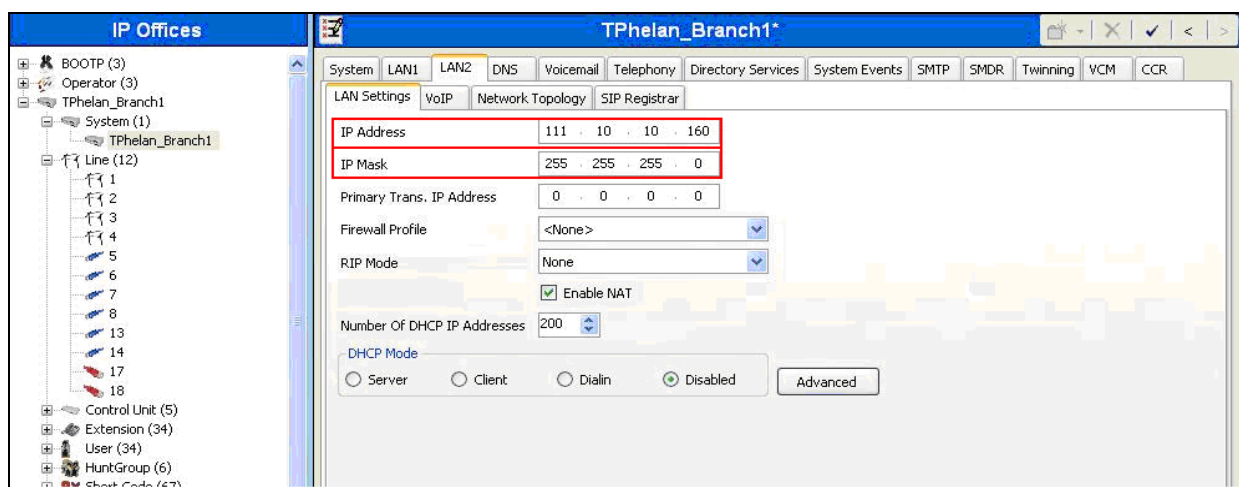
5.1. Verify System Capacity

Navigate to **License** → **SIP Trunk Channels** in the Navigation Pane. In the Details Pane verify that the License Status is Valid and that the number of Instances is sufficient to support the number of SIP trunk channels provisioned by Gamma Telecom.

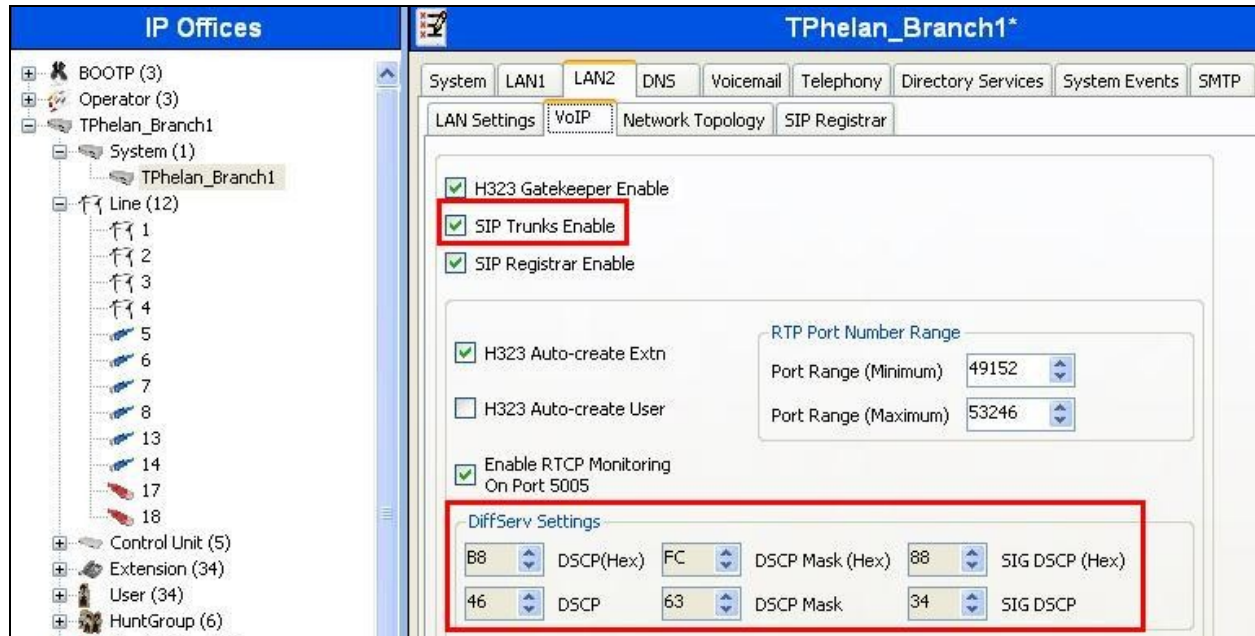


5.2. LAN2 Settings

In the sample configuration, the LAN2 port was used to connect the IP Office to the external intranet. To access the LAN2 settings, first navigate to **System** → **TPhelan_Branch1** in the Navigation Pane where TPhelan_Branch1 is the name of the IP Office. Navigate to the **LAN2** → **LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are set from values shown in **Figure 1**. All other parameters should be set according to customer requirements.

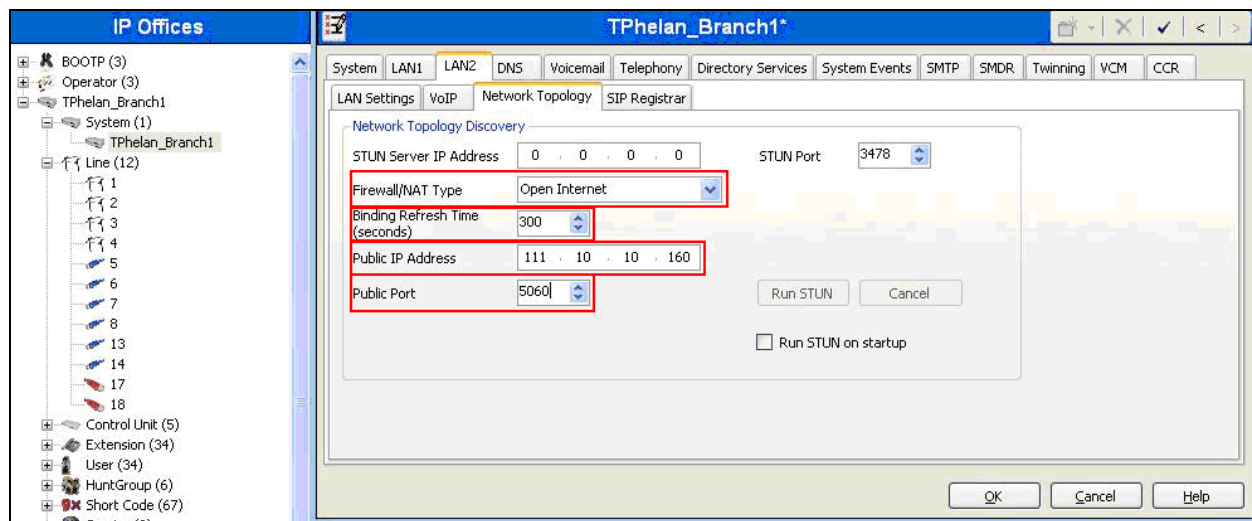


On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.



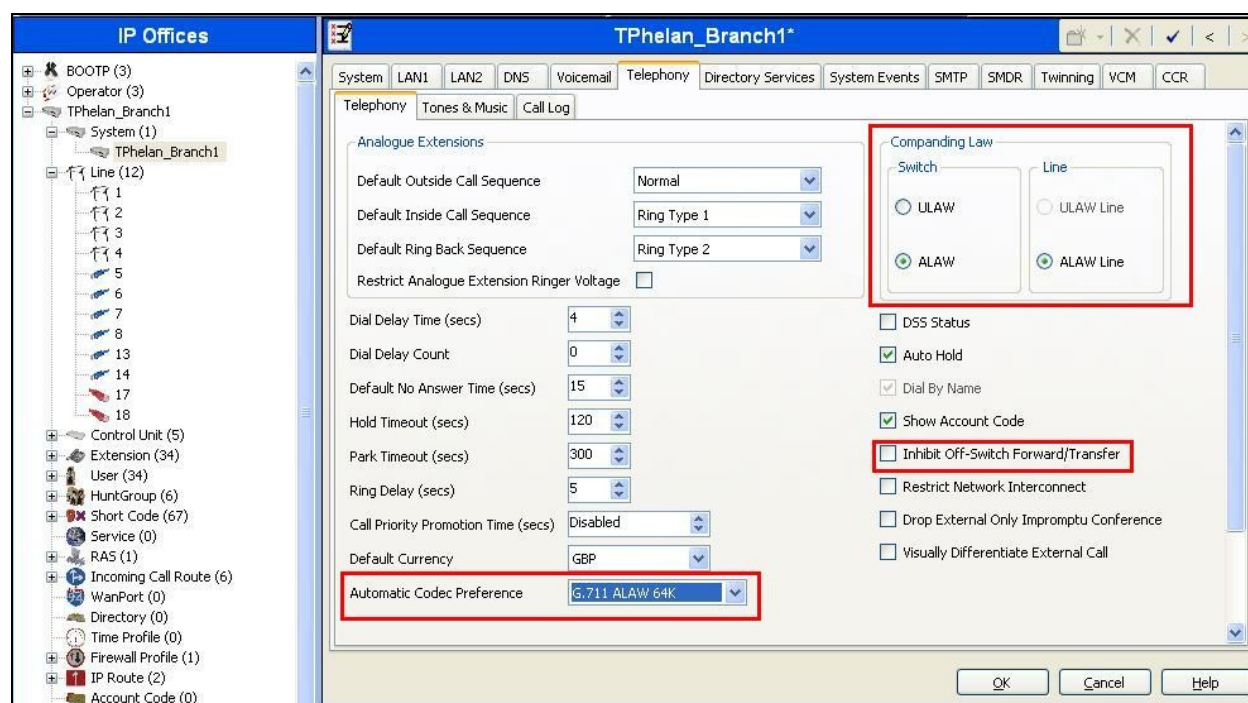
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**.
- Set **Binding Refresh Time (seconds)** to **300**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of LAN2.
- Set **Public Port** to **5060**.
- All other parameters should be set according to customer requirements.



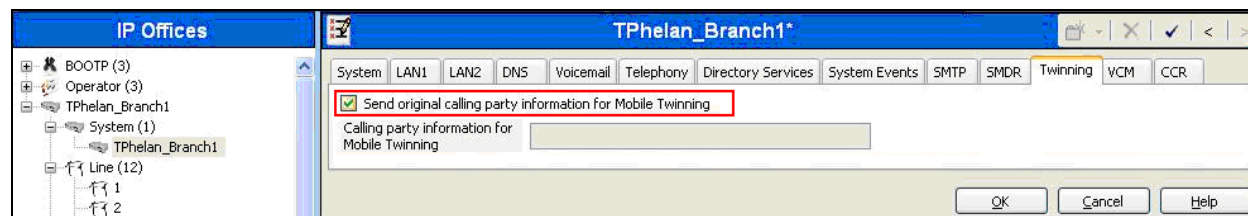
5.3. System Telephony Settings

Navigate to the **Telephony**→**Telephony** Tab on the Details Pane. Set the **Automatic Codec Preference** to the default codec to be used for intra-enterprise traffic. Choose the **Companding Law** typical for the enterprise location. For Europe, **ALAW** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk.



5.4. System Twinning Settings

Navigate to the **Twinning** tab, check the box labeled **Send original calling party information for Mobile Twinning**. With this setting, IP Office will send the original calling party number to the twinned phone in the SIP From header (not the associated desk phone number) for calls that originate from an internal extension. For inbound PSTN calls to a twinned enabled phone, IP Office will continue to send the associated host phone number in the SIP From header (used for the caller display). This setting only affects twinning and does not impact the messaging of other redirected calls such as forwarded calls. If this box is checked, it will also override any setting of the **Send Caller ID** parameter on the SIP line (**Section 5.5**).



5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between IP Office and the Gamma Telecom DirectConnect SIP Trunking service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New→SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters below to connect to the SIP Trunking service.

- Set the **ITSP Domain Name** to the domain name provided by Gamma Telecom (In this case an IP address was used).
- Set **Send Caller ID** to **None**. This parameter determines how the calling party number is sent in the SIP messaging for twinning if the box labeled **Send original calling party information for Mobile Twinning** is unchecked in **Section 5.4**. This parameter was set to **None** and the box in **Section 5.4** was checked.
- Ensure the **In Service** box is checked.
- Default values may be used for all other parameters.

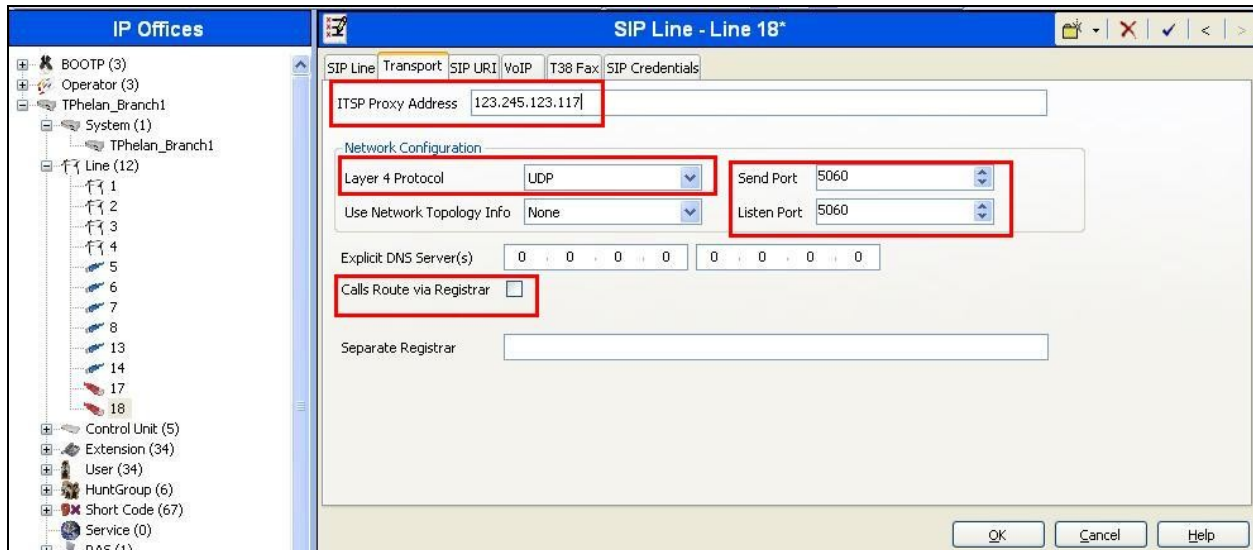
The screenshot shows the 'SIP Line - Line 18*' configuration window in IP Office. The left pane shows the 'IP Offices' tree with 'Line (12)' selected. The right pane shows the 'SIP Line' tab with the following configuration:

Parameter	Value
Line Number	18
ITSP Domain Name	123.245.123.117
In Service	<input checked="" type="checkbox"/>
Prefix	
National Prefix	0
Country Code	
International Prefix	00
Send Caller ID	None
REFER Support	<input checked="" type="checkbox"/>
Incoming	Auto
Outgoing	Auto
Use Tel URI	<input type="checkbox"/>
Check OOS	<input checked="" type="checkbox"/>
Call Routing Method	Request URI
Originator number for forwarded and twinning calls	

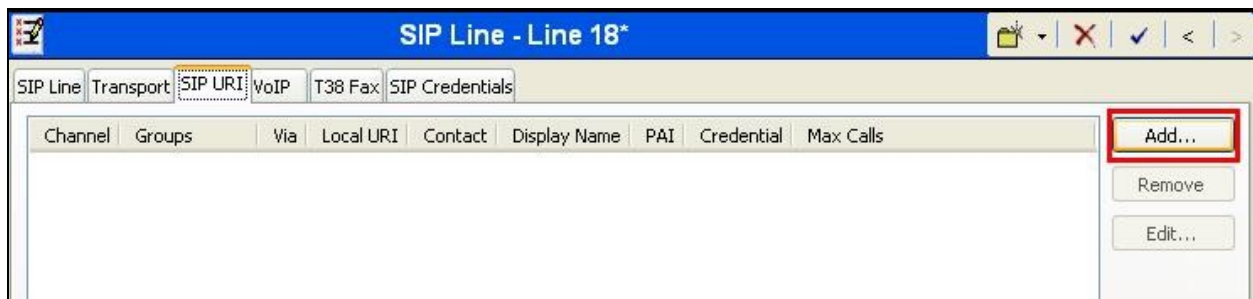
Buttons at the bottom: OK, Cancel, Help.

Select the **Transport** Tab and set the following:

- Set **ITSP IP Address** to the IP address of the Gamma Telecom DirectConnect SIP proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Uncheck **Calls Route via Registrar**.
- Set **Send Port** and **Listen Port** to **5060**.



After the SIP line parameters are defined, each SIP URI that IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane.



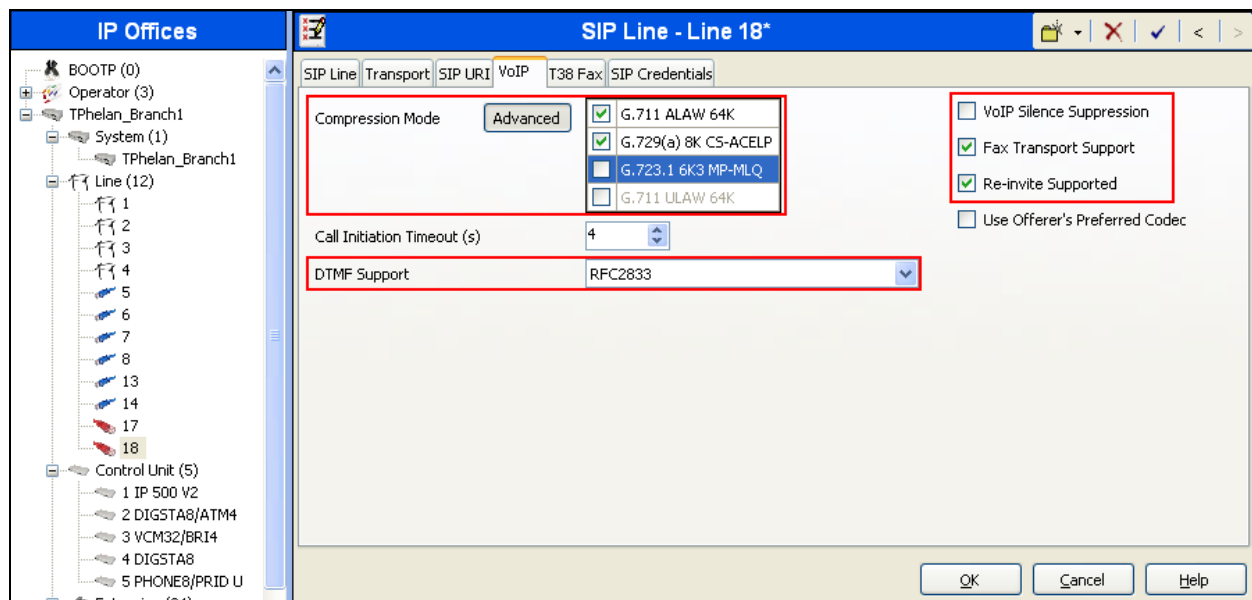
For the compliance test, a single SIP URI entry was created that matched any number assigned to an IP Office user. The entry was created with the parameters shown below.

- Set **Local URI**, **Contact**, **Display Name** and **PAI** to **Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.7**.
- For **Registration**, select **0: <None>** from the pull-down menu since this configuration does not use SIP registration.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **18** was defined that was associated to a single line (line 18).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

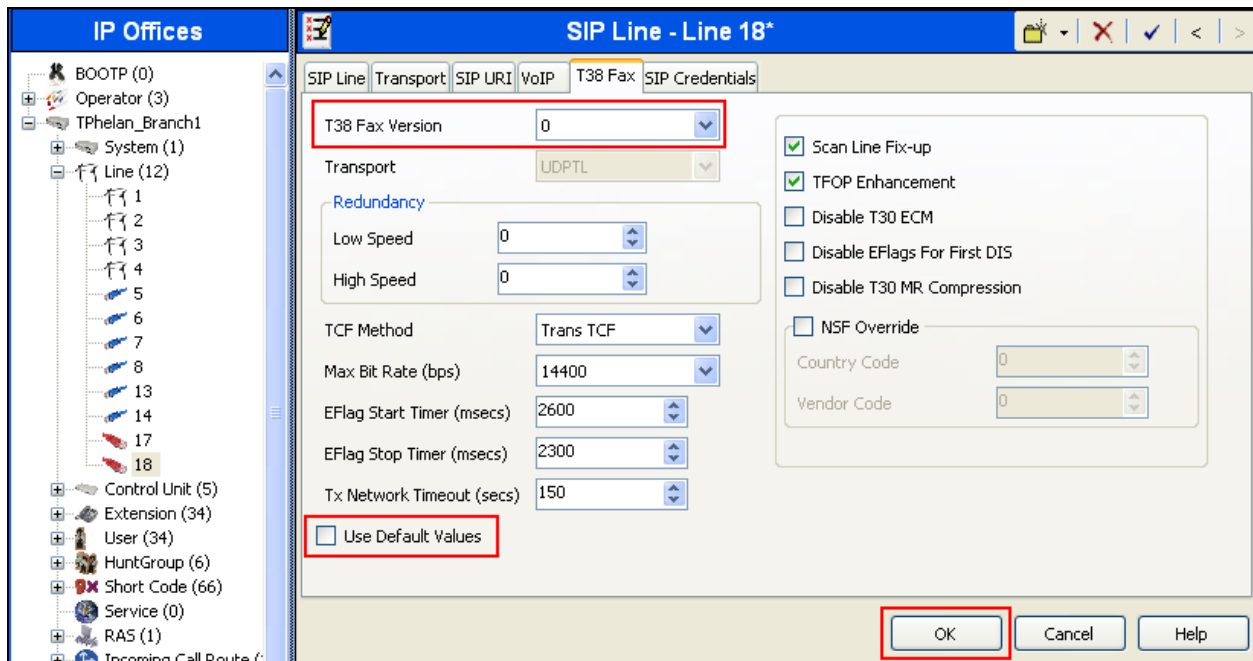
The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view shows the hierarchy: IP Offices, BOOTP (3), Operator (3), TPhelan_Branch1, System (1), TPhelan_Branch1, and Line (12). The 'Line (12)' folder is expanded, showing lines 1 through 18. Line 18 is selected. The main window is titled 'SIP Line - Line 18' and has tabs for SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'SIP URI' tab is active. It contains a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. The 'Edit Channel' dialog box is open, showing the configuration for line 18. The 'Via' field is set to '<None>'. The 'Local URI', 'Contact', 'Display Name', and 'PAI' fields are all set to 'Use Internal Data'. The 'Registration' field is set to '0: <None>'. The 'Incoming Group' and 'Outgoing Group' fields are both set to '18'. The 'Max Calls per Channel' field is set to '10'. The 'OK', 'Cancel', and 'Help' buttons are at the bottom right of the dialog box.

Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- Configure the **Compression Mode** with the **Advanced** button to specify the preferred order of the offered codecs. Select the codecs and their order based on the needs of the customer. Click the box next to the codec with the highest preference first, followed by the second preference. For the compliance test, **G.711 ALAW 64K** was selected first followed by **G.729(a) 8K CS-ACELP**.
- Set the **DTMF Support** field to **RFC2833**. This directs IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Fax Transport Support** box to allow T.38 fax operation.
- Check the **Re-invite Supported** box, to allow for codec re-negotiation in cases where the target of the incoming call or transfer does not support the codec originally negotiated on the trunk.
- Default values may be used for all other parameters.



Select the **T.38 Fax** tab, to set the T.38 parameters for the line. Un-check the **Use Default Values** box and select **0** from the **T38 Fax Version** drop down menu, all other field may retain their default values. Click the **OK** button to complete the SIP line configuration.



Note: It is advisable at this stage to save the configuration as described in **Section 5.10** to make the Line Group ID available in **Section 5.6**.

5.6. Short Codes

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@123.245.123.117"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.5**. This short code will use this line group when placing the outbound call.

Click the **OK** button.

The screenshot shows a software interface for configuring a short code. On the left is a navigation pane titled 'IP Offices' containing a list of entries such as *53*N#, *57*N#, *66, *68, *69, *70*N#, *71*N#, *80*N#, *81*N#, *9000*, *91N;, *92N;, *D55N, *SDN, *SKN, 200xx, 340xx, 350xx, 500xx, 777777777N, and 9N;. Below this list are icons for Service (0), RAS (1), Incoming Call Route (6), and WanPort (0). The main area is a 'Short Code' configuration window with a title bar '9N;; Dial'. It contains several fields: 'Code' (9N;), 'Feature' (Dial), 'Telephone Number' (N"@123.245.123.117"), 'Line Group Id' (18), 'Locale' (empty), and 'Force Account Code' (unchecked). At the bottom right are 'OK', 'Cancel', and 'Help' buttons. A red rectangle highlights the 'Code', 'Feature', 'Telephone Number', and 'Line Group Id' fields.

5.7. User

Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.5**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **SIP** tab in the Details Pane. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.5**). As such, these fields should be set to one of the DID numbers assigned to the enterprise from Gamma Telecom.

In the example below, the DID number **01555880762** is used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click the **OK** button.

Note: The **Contact** field must be in E.164 format for the caller ID on the called phone to display properly.

The screenshot shows the Avaya User configuration window. On the left is a tree view of IP Offices, with '89102 Extn89102' selected. The main pane is titled 'Extn89102: 89102' and has several tabs: 'Menu Programming', 'Mobility', 'Phone Manager Options', 'Hunt Group Membership', 'Announcements', 'SIP', and 'Personal Directory'. The 'SIP' tab is active, showing three text input fields: 'SIP Name' with the value '01555880762', 'SIP Display Name (Alias)' with the value 'Gamma 3', and 'Contact' with the value '+1555880762'. These three fields are enclosed in a red rectangular box. Below these fields is an unchecked checkbox labeled 'Anonymous'. At the bottom right of the window are three buttons: 'OK', 'Cancel', and 'Help'.

5.8. Incoming Call Routing

An incoming call route maps an inbound DID number on a specific line to an internal extension. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.5**.
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left.
- Default values can be used for all other fields.

The screenshot shows the 'IP Offices' interface. On the left is a navigation pane with a tree structure: BOOTP (3), Operator (3), TPhelan_Branch1, System (1), Line (12), Control Unit (5), Extension (34), User (34), HuntGroup (6), Short Code (67), Service (0), RAS (1), and Incoming Call Route (7). The 'Incoming Call Route (7)' is selected, showing a list of routes: 0, 18 01555880760, 18 01555880761, 18 01555880762, and 18 01555880763. The main pane shows the details for '18 01555880762' on the 'Standard' tab. The fields are: Bearer Capacity (Any Voice), Line Group Id (18), Incoming Number (01555880762), Incoming Sub Address (empty), Incoming CLI (empty), Locale (empty), Priority (1 - Low), Tag (empty), and Hold Music Source (System Source). A red box highlights the Bearer Capacity, Line Group Id, and Incoming Number fields.

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 01555880762 on line 18 are routed to extension 89102.

The screenshot shows the 'Destinations' tab for the same incoming call route. It displays a table with three columns: TimeProfile, Destination, and Fallback Extension. The 'TimeProfile' column has a dropdown menu with 'Default Value' selected. The 'Destination' column has a list of extensions: 89102 Extn89102, 89300 VoiceMail SIP, 8004 Extn8004, 89301 Voicemail SIP1, 89302 Voicemail SIP2, 89303 Voicemail SIP3, 89200 DDI 89200, and 89201 DDI 89201. The 'Fallback Extension' column has a dropdown menu with '89102 Extn89102' selected.

5.9. SIP Options

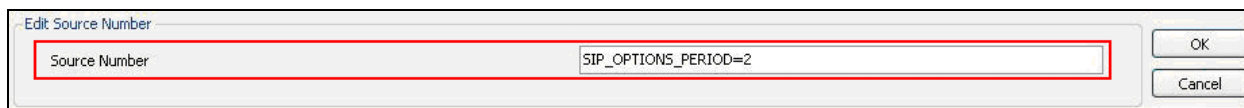
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user. The OPTIONS period is determined in the following manner:

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

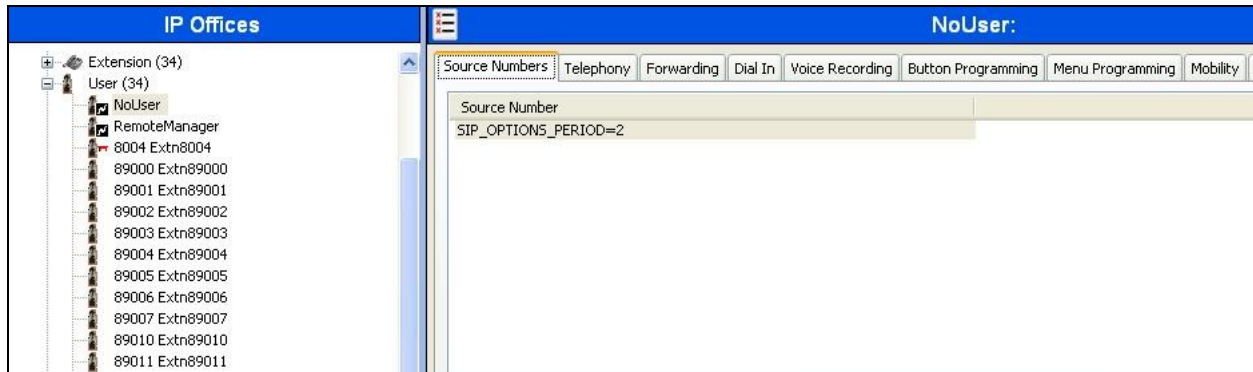
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User→NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the subsequent Details Pane, the **Source Number** field will appear. Enter **SIP_OPTIONS_PERIOD=X**, where **X** is the desired value in minutes. Click **OK**.



The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 2 minutes was desired. The **Binding Refresh Time** was set to **300** seconds (5 minutes) in **Section 5.2**. The **SIP_OPTIONS_PERIOD** was set to **2** minutes. Avaya IP Office chooses the OPTIONS period as the smaller of these two values (2 minutes). Click the **OK** button (not shown).



5.10. Save Configuration

Navigate to **File→Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

6. Gamma Telecom DirectConnect Service Configuration

Gamma Telecom is responsible for the configuration of the DirectConnect SIP Trunking service. The customer will need to provide the public IP address used to reach the IP Office at the enterprise. Gamma Telecom will provide the customer the necessary information to configure the SIP connection to the SIP Trunking service including:

- IP address of SIP Trunking SIP proxy
- Network SIP Domain
- Supported codecs
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

This section includes steps that can be used to verify that the configuration has been done correctly.

- Verify that the local extensions on IP Office can call and talk to each other.
- Verify that the IP Office Server can ping the Gamma Telecom Session Border Controller.
- Verify that the local extensions on IP Office and the telephones attached to the PSTN can call each other.
- Verify that it is possible to send FAX messages between the locally attached FAX device and the FAX unit attached to the PSTN.

8. Conclusion

The Gamma Telecom DirectConnect SIP Trunking service passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and the Gamma Telecom DirectConnect SIP Trunking service as shown in **Figure 1**.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and demonstrates Avaya IP Office can be configured to interoperate successfully with the Gamma Telecom IP DirectConnect Service. This solution provides Avaya IP Office users the ability to access the Public Switched Telephone Network (PSTN) via a SIP trunk using the Gamma Telecom IP Direct Connect Service.

9. Additional References

- [1] IP Office 6.1 Documentation CD, December 2010.
- [2] IP Office Installation, Document number15-601042, December 2010.
- [3] IP Office Manager, Document number15-601011, December 2010.
- [4] System Status Application, Document number15-601758, February 2010.

Product documentation for Avaya products may be found at <http://support.avaya.com>.

Product documentation for the Gamma Telecom DirectConnect service is available from Gamma Telecom.

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