



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Configuring MTS Allstream SIP Trunking with Avaya IP Office R9.0 - Issue 1.0

### Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between MTS Allstream and Avaya IP Office R9.0.

MTS Allstream SIP Trunking provides PSTN access via a SIP trunk between the enterprise and the MTS Allstream network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

MTS Allstream 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 MTS Allstream and Avaya IP Office R9.0.

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500V2 running Release 9.0 software, Avaya Voicemail Pro messaging application, Avaya H.323 and SIP hard phones, and SIP-based Avaya softphones.

The MTS Allstream SIP Trunking service provides PSTN access via a SIP trunk between the business site and the MTS Allstream 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

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.

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to the MTS Allstream SIP Trunking service via the public Internet. The configuration shown in **Figure 1** was used to exercise the features and functionality tests listed in **Section 2.1**

### 2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the compliance test.

- SIP OPTIONS queries and responses.
- Incoming PSTN calls to H.323 and SIP telephones at the business site. All inbound PSTN calls were routed to the business site across the SIP trunk from the service provider.
- Outgoing PSTN calls from H.323 and SIP telephones at the business site. All outbound PSTN calls were routed from the business site across the SIP trunk to the service provider.
- Various call types including: local, long distance, outbound toll-free, international, Operator (0), Operator-Assisted (0 + 10-digits) and directory assistance.
- G.729A and G.711MU codecs.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail access and navigation for inbound and outbound calls.
- Telephony supplementary features such as hold and resume, transfer, and conference.
- Off-net call forwarding and call transfer/conference.
- Twinning to PSTN mobile phones on inbound calls.
- Use of SIP INVITE message for call redirection to the PSTN.

- Inbound and outbound long-duration calls stability.
- Inbound and outbound long holding time call stability.
- Response to incomplete call attempts and trunk busy or error conditions.
- T.38 fax.

Items not supported or not tested include the following:

- Inbound toll-free and emergency calls (911) were not tested as part of the compliance test.
- MTS Allstream SIP Trunking does not support use of the SIP REFER method for network redirection (transferring calls with the PSTN back to the PSTN).

## 2.2. Test Results

Interoperability compliance testing of MTS Allstream SIP Trunking was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Codec Lockdown on Outbound Calls** – When Avaya IP Office was configured with the G.729A and G.711MU codecs in that preference order, MTS Allstream responded to the outbound INVITE request with both codecs in the SDP (with G.729A listed first) instead of selecting one from the INVITE SDP list. This behavior had no user impact. Calls were successful using G.729A.
- **Codec Preference** – When Avaya IP Office was configured with the same 2 codecs as above but with G.711MU as the preferred codec, MTS Allstream responded to the outbound INVITE request with the same 2 codecs in the SDP, but with G.729A listed first. Calls were successful using G.729A instead of G.711MU.
- **Direct Media** – Avaya IP Office R9.0 offers a new Direct Media capability on IP Office 500V2 that allows IP endpoints to send RTP media directly to each other rather than having all the media flow through the IP Office, using up VoIP and relay resources. This capability is not supported on the configuration described in these application notes where the Avaya IP Office and IP phones are connected to a private network and the Avaya IP Office connects to the service provider network via a direct public Internet connection without using an enterprise Session Border Controller.

## 2.3. Support

For technical support on the MTS Allstream SIP Trunking service, the MTS Allstream customer support web page at <https://www.mts.ca/mts/contact+us> contains the phone and email access information.

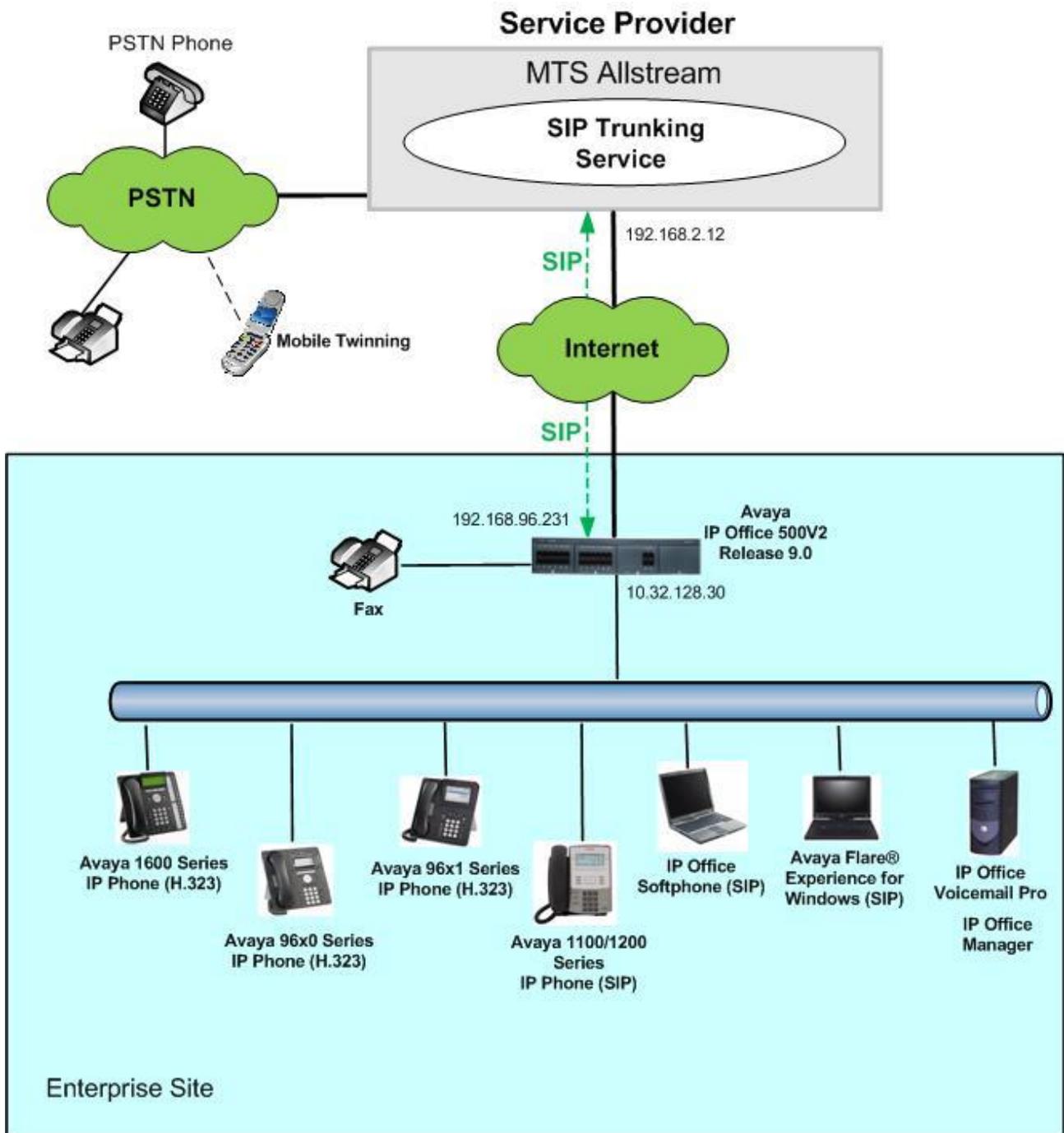
Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

### 3. Reference Configuration

**Figure 1** illustrates the test configuration showing a business site connected to MTS Allstream SIP Trunking.

Located at the business site is an Avaya IP Office 500V2. Endpoints include various Avaya IP Telephones (with H.323 and SIP firmware) and SIP-based Avaya softphones (Avaya IP Office Softphone and Avaya Flare® Experience for Windows). The site also has a Windows PC running Avaya Voicemail Pro for providing voice messaging service to the Avaya IP Office users, and Avaya IP Office Manager for administering the Avaya IP Office.

Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user phones will also ring and can be answered at the configured mobile phones.



**Figure 1: Test Configuration**

For security purposes, any actual public IP addresses used in the compliance test were changed to 192.168.x.x throughout these Application Notes.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to MTS Allstream. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent to the service provider network. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits for long distance calls and local calls. Thus, for these NANP calls, Avaya IP Office sent 11 digits in the Request URI and the To header of an outbound SIP INVITE message. MTS Allstream sent 10 digits in the Request URI and the To header of inbound SIP INVITE messages.

In an actual customer configuration, the business site may also include additional network components between the service provider and Avaya 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 Avaya IP Office must be allowed to pass through these devices.

## 4. Equipment and Software Validated

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

<b>Avaya Telephony Components</b>	
<b>Equipment / Software</b>	<b>Release / Version</b>
Avaya IP Office 500V2	9.0.0.0 Build 829
Avaya IP Office COMBO6210/ATM4 Module	9.0.0.0 Build 829
Avaya IP Office Manager	9.0.0.0 Build 829
Avaya Voicemail Pro	9.0.0.311
Avaya 1616 IP Telephones (H.323)	Avaya one-X Deskphone 1.3 SP3
Avaya 9611G IP Telephones (H.323)	Avaya one-X Deskphone 6.3
Avaya 9630G IP Telephones (H.323)	Avaya one-X Deskphone 3.2
Avaya 1120E IP Telephone (SIP)	4.03.18.00
Avaya 1140E IP Telephone (SIP)	4.03.18.00
Avaya IP Office Softphone	3.2.3.49 68975
Avaya Flare® Experience for Windows	1.1.4.23
<b>MTS Allstream Components</b>	
<b>Equipment / Software</b>	<b>Release / Version</b>
Genband S3 Session Border Controller	5.2.2.12
Nortel CS2K	CVM13

## 5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running Avaya IP Office Manager, select **Start → Programs → IP Office → Manager** to launch the application. A screen that includes the following in the center may be displayed:

### **WELCOME to IP Office Administration**

#### **What would you like to do ?**

[Create an Offline Configuration](#)

[Open Configuration from System](#)

[Read a Configuration from File](#)

Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials.

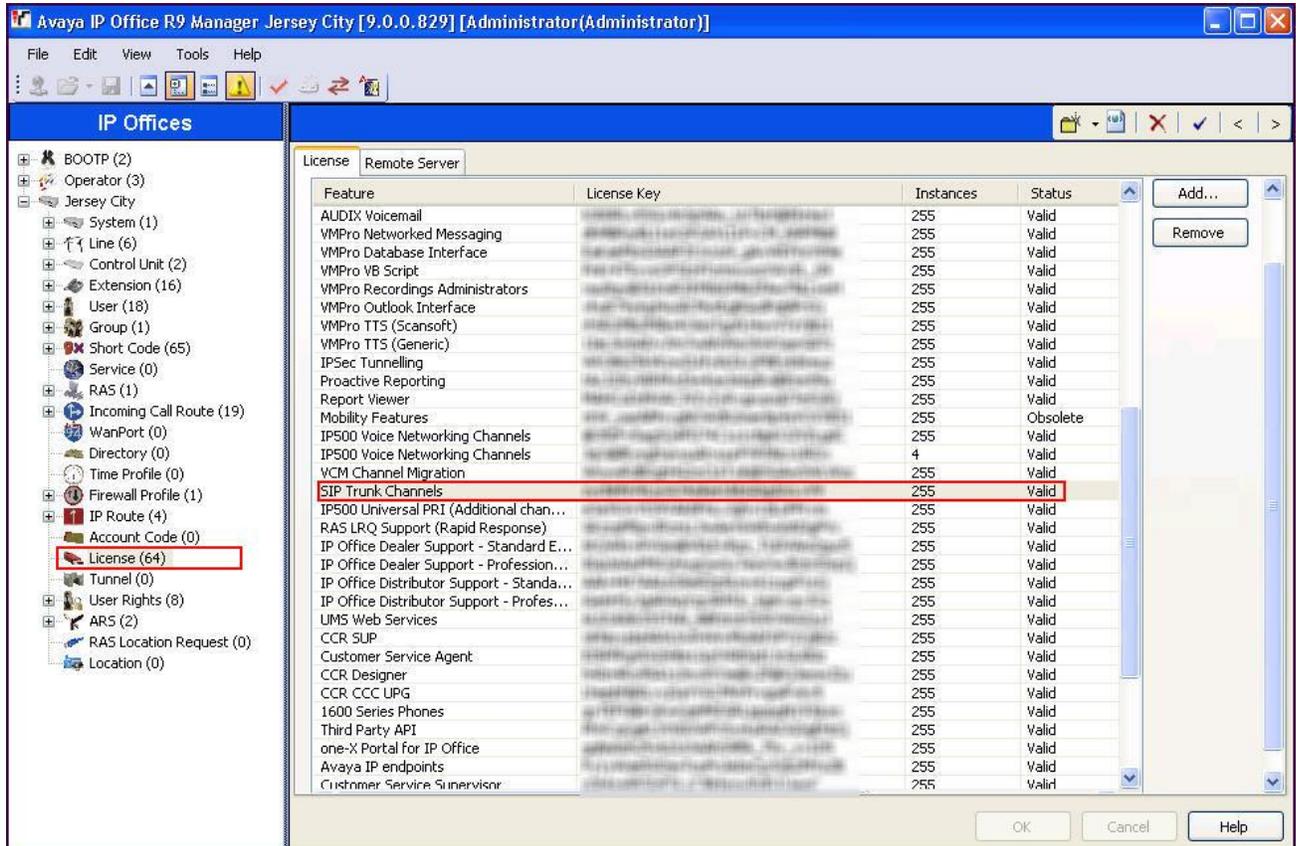
The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show the Details pane on the right side. Since the Group Pane has been omitted, its content is shown as submenus in the Navigation pane. These panes (Navigation and Details) will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and IP Office Softphone support) is assumed to already be in place.

In the sample configuration, **Jersey City** was used as the system name. All navigation described in the following sections (e.g., **License → SIP Trunk Channels**) appears as submenus underneath the system name **Jersey City** in the Navigation Pane. The configuration screens only show values/settings configured for the compliance test. Defaults were used for other values and may be customized based upon requirements in the field.

## 5.1. Licensing and Physical Hardware

The configuration and features described in these Application Notes require Avaya IP Office to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

To verify that there is a **SIP Trunk Channels** License with sufficient capacity; click **License** in the Navigation pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane.



The screenshot shows the Avaya IP Office R9 Manager interface. The left navigation pane is expanded to show the 'License' option under the 'Jersey City' configuration. The main window displays the 'License' configuration page for a 'Remote Server'. A table lists various features and their license details.

Feature	License Key	Instances	Status
AUDIX Voicemail		255	Valid
VMPPro Networked Messaging		255	Valid
VMPPro Database Interface		255	Valid
VMPPro VB Script		255	Valid
VMPPro Recordings Administrators		255	Valid
VMPPro Outlook Interface		255	Valid
VMPPro TTS (Scansoft)		255	Valid
VMPPro TTS (Generic)		255	Valid
IPSec Tunnelling		255	Valid
Proactive Reporting		255	Valid
Report Viewer		255	Valid
Mobility Features		255	Obsolete
IP500 Voice Networking Channels		255	Valid
IP500 Voice Networking Channels		4	Valid
VCM Channel Migration		255	Valid
<b>SIP Trunk Channels</b>		<b>255</b>	<b>Valid</b>
IP500 Universal PRI (Additional chan...		255	Valid
RAS LRQ Support (Rapid Response)		255	Valid
IP Office Dealer Support - Standard E...		255	Valid
IP Office Dealer Support - Profession...		255	Valid
IP Office Distributor Support - Standa...		255	Valid
IP Office Distributor Support - Profes...		255	Valid
UMS Web Services		255	Valid
CCR SUP		255	Valid
Customer Service Agent		255	Valid
CCR Designer		255	Valid
CCR CCC UPG		255	Valid
1600 Series Phones		255	Valid
Third Party API		255	Valid
one-X Portal for IP Office		255	Valid
Avaya IP endpoints		255	Valid
Customer Service Supervisor		255	Valid

To view the physical hardware comprising the Avaya IP Office system, expand the components under the **Control Unit** in the Navigation pane. In the sample configuration, the second component listed is a Combination Card. This module has 6 digital station ports, two analog extension ports, 4 analog trunk ports and 10 VCM channels. The VCM is a Voice Compression Module supporting VoIP codecs. An Avaya IP Office hardware configuration with a VCM component is necessary to support SIP trunking.

To view the details of the component, select the component in the Navigation pane.

The screen below shows the details of the IP 500 V2:

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, and on the right is the 'IP 500 V2' details pane. The navigation pane shows a tree structure with 'Control Unit (2)' expanded to show '1 IP 500 V2' selected. The details pane shows the following information:

Unit	
Device Number	1
Unit Type	IP 500 V2
Version	9.0.0.829
Serial Number	
Unit IP Address	10.32.128.30
Interconnect Number	0
Module Number	Control Unit

The screen below shows the details of the Combination Card:

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, and on the right is the 'COMBO6210/ATM4' details pane. The navigation pane shows a tree structure with 'Control Unit (2)' expanded to show '2 COMBO6210/ATM4' selected. The details pane shows the following information:

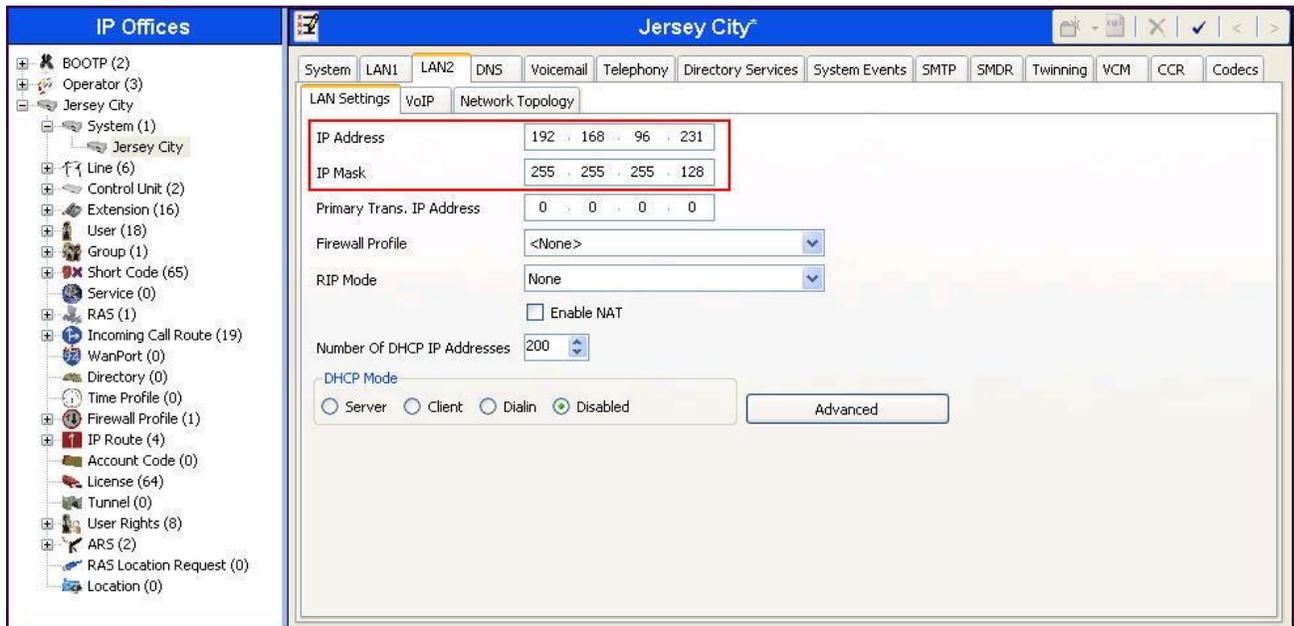
Unit	
Device Number	2
Unit Type	COMBO6210/ATM4
Version	9.0.0.829
Serial Number	
Unit IP Address	0.0.0.0
Interconnect Number	0
Module Number	Control Unit

## 5.2. System

This section configures the necessary system settings

### 5.2.1. LAN2 Tab

In the sample configuration, *Jersey City* was used as the system name and the WAN port (LAN2 port) was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN interface on Avaya IP Office. To access the LAN2 settings, first navigate to **Jersey City** → **System** → **Jersey City** in the Navigation Pane and then navigate to the **LAN2** → **LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the mask used on the public network.



The screenshot displays the Avaya IP Office configuration interface for the system named "Jersey City". The left-hand side shows a navigation tree under "IP Offices" with "Jersey City" selected. The main pane shows the "LAN2" tab, with the "LAN Settings" sub-tab active. The "IP Address" field is set to "192 . 168 . 96 . 231" and the "IP Mask" field is set to "255 . 255 . 255 . 128". Other settings include "Primary Trans. IP Address" set to "0 . 0 . 0 . 0", "Firewall Profile" set to "<None>", "RIP Mode" set to "None", and "Number Of DHCP IP Addresses" set to "200". The "DHCP Mode" is set to "Disabled".

Field	Value
IP Address	192 . 168 . 96 . 231
IP Mask	255 . 255 . 255 . 128
Primary Trans. IP Address	0 . 0 . 0 . 0
Firewall Profile	<None>
RIP Mode	None
Number Of DHCP IP Addresses	200
DHCP Mode	Disabled

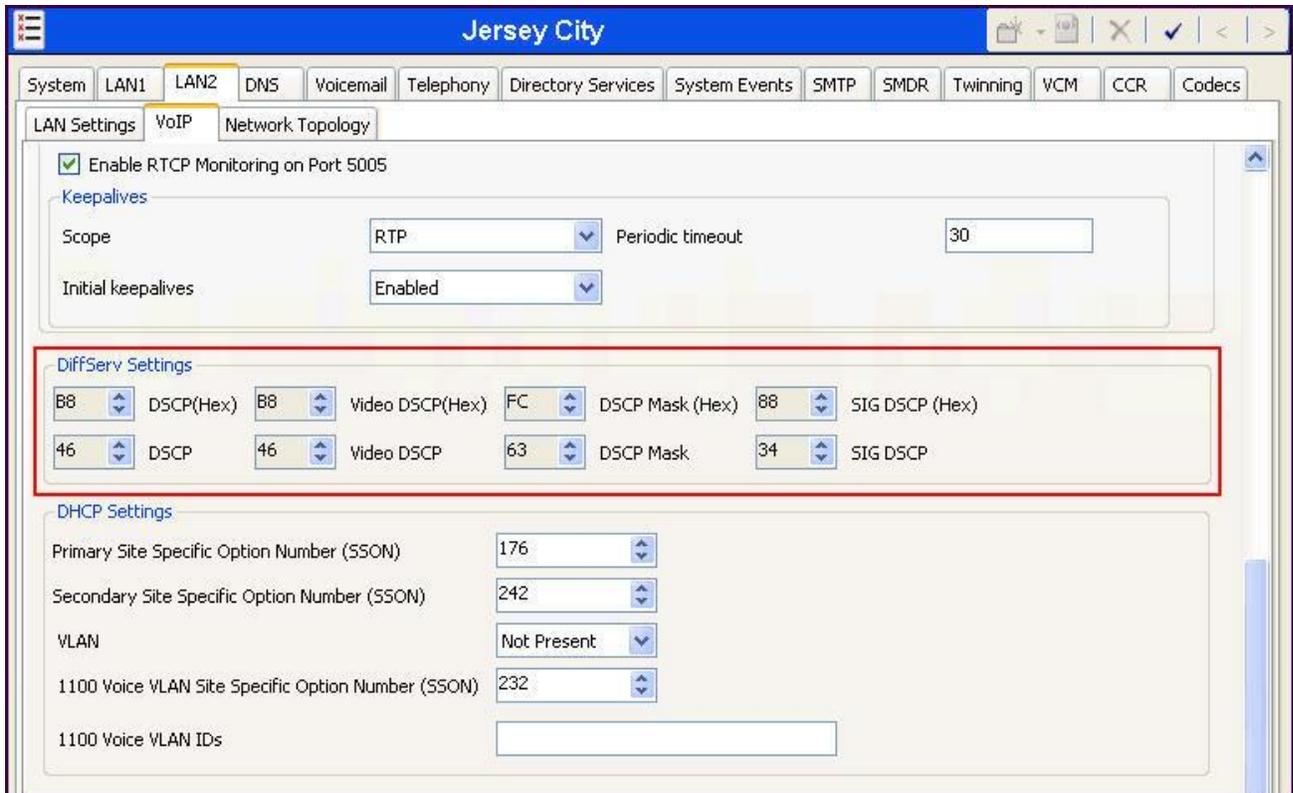
On the **VoIP** tab of LAN2 in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks.

In the **RTP** area:

- 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 port in the configurable range for calls using LAN2.
- In the **Keepalives** section. Select **RTP** for **Scope**; select **Enabled** for **Initial keepalives**; enter **30** for **Periodic timeout**. These settings direct Avaya IP Office to send artificial RTP packets toward the service provider at the start of the call to prevent audio loss in certain off-net call redirection scenarios. Some service providers expect the IP Office endpoint to send RTP packets first even though there is no IP Office media endpoint involved in this call situation since the call has been re-directed back to the PSTN

The screenshot shows the configuration interface for Jersey City, specifically the VoIP tab for LAN2. The 'SIP Trunks Enable' checkbox is checked and highlighted with a red box. Below it, the 'RTP' section is visible, with 'Port Number Range' and 'Keepalives' sections also highlighted with red boxes. The 'Port Number Range' section shows Minimum: 49152 and Maximum: 53246. The 'Keepalives' section shows Scope: RTP, Initial keepalives: Enabled, and Periodic timeout: 30.

Scroll down to the **DiffServ Settings** section. Avaya IP Office can 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.



On the **Network Topology** tab of LAN2 in the Details Pane, configure the following parameters:

- Select **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**. With the **Open Internet** setting, **STUN Server IP Address** is not used.
- Set **Binding Refresh Time (seconds)** to a desired value. 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. See **Section 5.8** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set **Public Port** to **5060** for **UDP**.

The screenshot shows the 'Jersey City' network management interface. The 'Network Topology' tab is active for LAN2. The 'Network Topology Discovery' section contains the following settings:

- STUN Server Address: 69.90.168.13
- STUN Port: 3478
- Firewall/NAT Type: Open Internet (highlighted with a red box)
- Binding Refresh Time (seconds): 60
- Public IP Address: 192 . 168 . 96 . 231

Below these settings are 'Run STUN' and 'Cancel' buttons. The 'Public Port' section includes:

- UDP: 5060 (highlighted with a red box)
- TCP: 0
- TLS: 0

At the bottom, there is a checkbox for 'Run STUN on startup' which is currently unchecked.

During the compliance testing, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with MTS Allstream, and therefore is not described in these Application Notes.

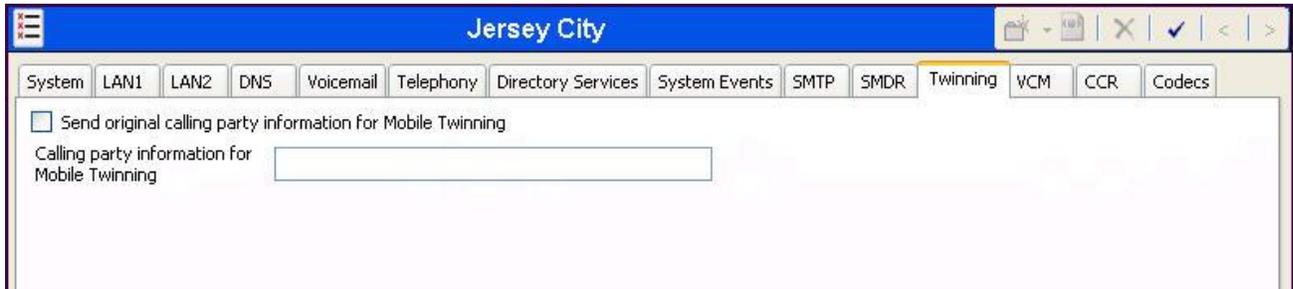
## 5.2.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** tab on the Details Pane. Enter or select **0** for **Hold Timeout (secs)** so that calls on hold will not time out. Choose the **Companding Law** typical for the business site. For the compliance test, **U-LAW** was 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 per customer business policies.

The screenshot displays the 'Jersey City' configuration interface for the 'Telephony' tab. The 'Hold Timeout (secs)' field is highlighted with a red box and set to 0. The 'Companding Law' section is also highlighted with a red box, showing 'U-Law' selected for both 'Switch' and 'Line'. The 'Inhibit Off-Switch Forward/Transfer' checkbox is unchecked and highlighted with a red box. Other settings include 'Dial Delay Time (secs)' at 4, 'Dial Delay Count' at 0, 'Default No Answer Time (secs)' at 15, 'Park Timeout (secs)' at 300, 'Ring Delay (secs)' at 5, 'Call Priority Promotion Time (secs)' at Disabled, 'Default Currency' at USD, 'Default Name Priority' at Favor Trunk, and 'Media Connection Preservation' at Disabled. The 'Auto Hold', 'Dial By Name', 'Show Account Code', 'High Quality Conferencing', and 'Digital/Analogue Auto Create User' checkboxes are checked.

### 5.2.3. Twinning Calling Party Settings

To view or change the System Twinning settings, navigate to the **Twinning** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank.



### 5.3. IP Route

Navigate to **IP Route → 0.0.0.0** in the left Navigation Pane if a default route already exists. Otherwise, to create the default route, right-click on **IP Route** and select **New** (not shown). Create/verify a default route with the following parameters:

- Set **IP Address** and **IP Mask** to **0.0.0.0**.
- Set **Gateway IP Address** to the IP address of the gateway for the public internet WAN network.
- Set **Destination** to **LAN2** from the drop-down list.



## 5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the MTS Allstream SIP Trunking service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2**.

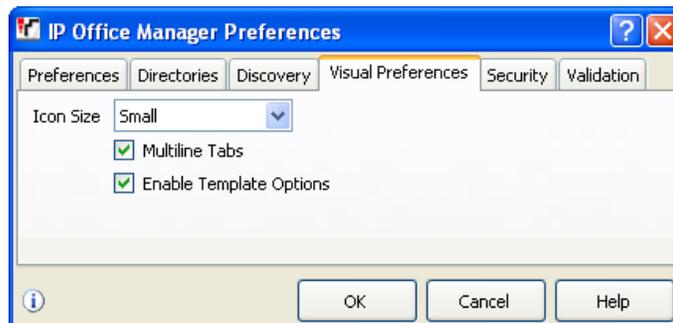
Also, the following SIP Line settings are not supported on Basic Edition:

- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

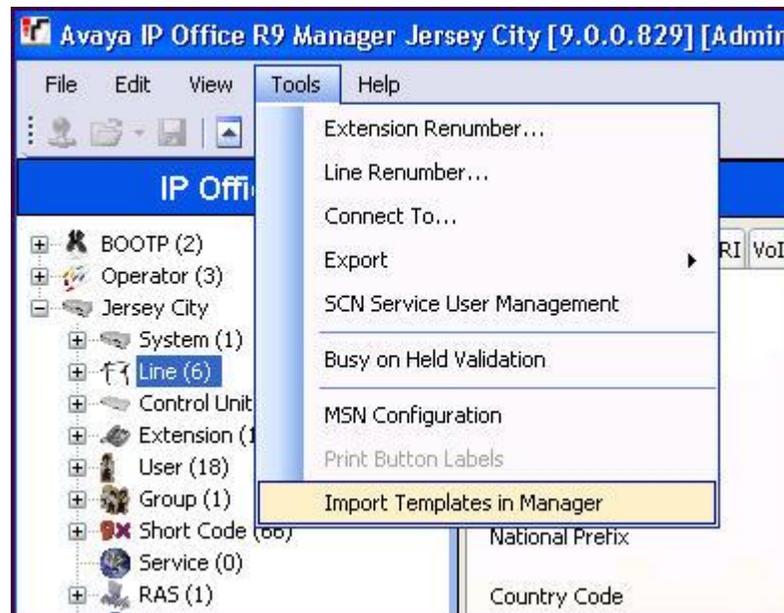
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.2**.

### 5.4.1. Create SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **US\_ MTS Allstream\_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.
2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Verify that the option box is checked next to **Enable Template Options**. Click **OK**.

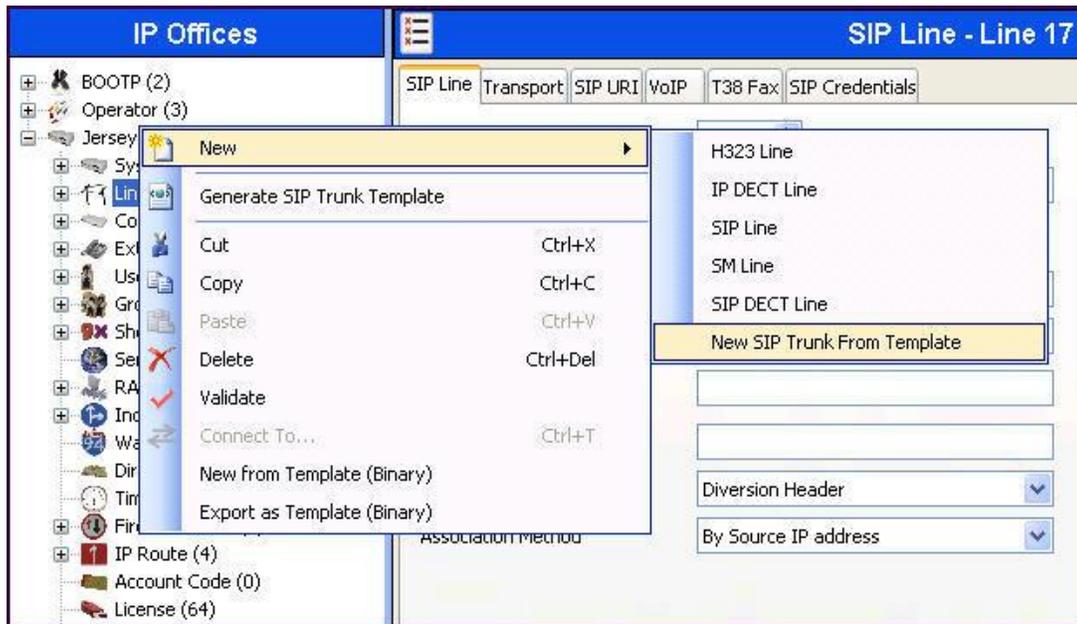


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools** → **Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

- To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New SIP Trunk From Template**.



- In the subsequent **Template Type Selection** pop-up window, select *United States* from the **Country** drop-down list and select *MTS Allstream* from the **Service Provider** drop-down list as shown below. These values correspond to parts of the file name (**US\_MTS Allstream\_SIPTrunk.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



Note that the newly created SIP Line may not immediately appear in the Navigation pane until the configuration was saved, closed and reopened in IP Office Manager.

- Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2**.

## 5.4.2. SIP Line Configuration - Manual

A SIP line is needed to establish the SIP connection between Avaya IP Office and MTS Allstream SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New** → **SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

- Set **ITSP Domain Name** to the LAN2 IP address (**192.168.96.231**) so that Avaya IP Office uses this IP as the URI host in SIP headers such as From and Diversion.
- Check the **In Service** box.
- Check **OOS** box. Avaya IP Office will use the SIP OPTIONS method to periodically check the SIP Line. See **Section 5.8** for details on time between SIP OPTIONS sent by IP Office.
- Set **Call Routing Method** to **Request URI**. Avaya IP Office will route inbound calls based on the number in the Request URI.
- Set **Send Caller ID** to **Diversion Header**. With this setting and the related configuration in **Section 5.2.3**, Avaya IP Office will include the Diversion Header for calls that are redirected via Mobile Twinning out the SIP Line to the service provider. It will also include the Diversion Header for calls that are forwarded out the SIP Line..
- Uncheck **REFER Support**. MTS Allstream SIP Trunking does not supports use of REFER for off-net call re-direction as in call transfer.
- Set **Method for Session Refresh** to **Auto**. With this setting Avaya IP Office will send UPDATE messages for session refresh if the other party supports UPDATE. If UPDATE is not supported, re-INVITE messages are sent.

The screenshot displays the configuration window for a SIP Line in Avaya IP Office. The left pane shows a tree view of the system hierarchy, with 'Line 17' selected. The main pane is titled 'SIP Line - Line 17\*' and contains the following configuration fields:

- Line Number:** 17
- ITSP Domain Name:** 192.168.96.231
- In Service:**
- URI Type:** SIP
- Prefix:** (empty)
- National Prefix:** (empty)
- Country Code:** (empty)
- International Prefix:** (empty)
- Send Caller ID:** Diversion Header
- Association Method:** By Source IP address
- Check OOS:**
- Call Routing Method:** Request URI
- Originator number for forwarded and twinning calls:** (empty)
- Name Priority:** System Default
- Caller ID from From header:**
- Send From In Clear:**
- User-Agent and Server Headers:** (empty)
- Service Busy Response:** 486 - Busy Here
- Action on CAC Location Limit:** Allow Voicemail
- REFER Support:** 
  - Incoming:** Always
  - Outgoing:** Always
- Method for Session Refresh:** Auto
- Session Timer (seconds):** On Demand
- Media Connection Preservation:** Disabled

Navigate to the **Transport** tab and set the following:

- Set the **ITSP Proxy Address** to the IP address of the service provider SIP Trunking access interface provided by MTS Allstream.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2** as configured in **Section 5.2.1**.
- Set the **Send Port** to **5060**.

The screenshot shows the 'SIP Line - Line 17\*' configuration window. The 'Transport' tab is selected. The 'ITSP Proxy Address' is set to '192.168.2.12'. The 'Network Configuration' section is highlighted with a red box and contains the following settings: 'Layer 4 Protocol' is 'UDP', 'Send Port' is '5060', 'Use Network Topology Info' is 'LAN 2', and 'Listen Port' is '5060'. Below this, 'Explicit DNS Server(s)' are set to '0.0.0.0'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

Select the **SIP URI** tab to create a SIP URI entry or edit an existing entry. A SIP URI entry matches each incoming number that Avaya IP Office will accept on this line. 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 to match any DID number assigned to an Avaya IP Office user. The following screen shows the edit window on a previously configured entry for the compliance test.

- Set **Local URI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **Contact** and **Display Name** to **Use Internal Data**. This setting will cause the Contact and Display Name data to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.68**.
- Set **PAI** to **Use Internal Data**. This setting directs Avaya IP Office to send the PAI (P-Asserted-Identity) header when appropriate. The PAI header will be populated from the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.68**.
- For **Registration**, select **0:<None>** from the pull-down menu. The test circuit used for the compliance test did not require trunk registration.

- Associate this line with an incoming line group by entering 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. For the compliance test, the incoming and outgoing group **17** was specified (Note: In the sample configuration, the line group number happened to be identical to the SIP Line number, but these two numbers do not need to be the same).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed using this SIP URI pattern.

The screenshot shows the 'SIP Line - Line 17\*' configuration window. At the top, there are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. Below the tabs is a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI. The table contains one row with Channel 1, Groups 17 17, and Via 1... To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is an 'Edit Channel' dialog box. The dialog box has the following fields: 'Via' (text box with '192.168.96.231'), 'Local URI' (dropdown menu with 'Use Internal Data'), 'Contact' (dropdown menu with 'Use Internal Data'), 'Display Name' (dropdown menu with 'Use Internal Data'), 'PAI' (dropdown menu with 'Use Internal Data'), 'Registration' (dropdown menu with '0: <None>'), 'Incoming Group' (text box with '17'), 'Outgoing Group' (text box with '17'), and 'Max Calls per Channel' (spin box with '10'). To the right of the dialog box are 'OK' and 'Cancel' buttons.

Select the **VoIP** tab. Set the parameters as shown below.

- Set the **Codec Selection** to *Custom*.
- Choose **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** from the **Unused** box and move these 2 selections to the **Selected** box. Use the down/up arrows to order the 2 selected codecs as shown. These 2 codecs are supported by MTS Allstream SIP Trunking.
- Select **T38** for **Fax Transport Support**.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones as out-band RTP events as per RFC2833.
- Uncheck the **VoIP Silence Suppression** option box.
- Check the **Re-invite Supported** option box.
- Check the **PRACK/100rel Supported** option box. This setting enables support by Avaya IP Office for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.

The screenshot displays the configuration window for 'SIP Line - Line 17'. The 'VoIP' tab is selected. The 'Codec Selection' dropdown is set to 'Custom'. Below this, there are two boxes: 'Unused' and 'Selected'. The 'Unused' box contains 'G.711 ALAW 64K' and 'G.723.1 6K3 MP-MLQ'. The 'Selected' box contains 'G.729(a) 8K CS-ACELP' and 'G.711 ULAW 64K'. To the right of these boxes are navigation arrows. Below the codec boxes, the 'Fax Transport Support' dropdown is set to 'T38'. The 'Location' dropdown is set to 'Cloud'. The 'Call Initiation Timeout (s)' is set to '4'. The 'DTMF Support' dropdown is set to 'RFC2833'. On the right side of the window, there are several checkboxes: 'VoIP Silence Suppression' (unchecked), 'Allow Direct Media Path' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'PRACK/100rel Supported' (checked), 'Force direct media with phones' (unchecked), and 'G.711 Fax ECAN' (unchecked). Red boxes highlight the 'Codec Selection' dropdown, the 'Unused' and 'Selected' boxes, the 'Fax Transport Support' dropdown, the 'DTMF Support' dropdown, and the 'Re-invite Supported' and 'PRACK/100rel Supported' checkboxes.

Select the **T38 Fax** tab to set the Fax over Internet Protocol parameters of the SIP line. Set the parameters as shown below.

- Uncheck **Use Default Values** at the bottom of the screen.
- Set **T38 Fax Version** to **0**. MTS Allstream SIP Trunking supports T.38 fax version 0.
- Set **Max Bit Rate (bps)** to 14400, the highest fax bit rate that Avaya IP Office supports for T.38 faxing.
- Check the **Disable T30 ECM** option.
- Default values may be used for all other parameters.

The screenshot shows the configuration window for 'SIP Line - Line 17'. The 'T38 Fax' tab is selected. The following parameters are visible and highlighted with red boxes:

- T38 Fax Version:** 0
- Max Bit Rate (bps):** 14400
- Disable T30 ECM:**
- Use Default Values:**

Other visible parameters include:

- Transport: UDPTL
- Redundancy: Low Speed (0), High Speed (0)
- TCF Method: Trans TCF
- EFlag Start Timer (msecs): 2600
- EFlag Stop Timer (msecs): 2300
- Tx Network Timeout (secs): 200
- Scan Line Fix-up:
- TFOP Enhancement:
- Disable EFlags For First DIS:
- Disable T30 MR Compression:
- NSF Override:
- Country Code: 0
- Vendor Code: 0

## 5.5. Short Code

Define a short code to route outbound calls to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). 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. The **9N;** short code, used for the compliance test, 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"@192.168.2.12"**. 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. The IP address following the @ sign is the IP address of the MTS Allstream SIP Trunking access interface.
- Set the **Line Group Id** to the **Outgoing Group** number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4.2**. This short code will use this line group when placing the outbound calls.

The screenshot displays the Avaya Management System interface. On the left is the 'IP Offices' navigation pane, showing a tree structure with categories like BOOTP (2), Operator (3), Jersey City, System (1), Line (6), Control Unit (2), Extension (16), User (18), Group (1), Short Code (65), Service (0), RAS (1), Incoming Call Route (19), WanPort (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (4), and Account Code (0). The main area is titled '9N;: Dial' and contains a 'Short Code' configuration form. The form fields are: Code (9N;), Feature (Dial), Telephone Number (N"@192.168.2.12"), Line Group ID (17), Locale (United States (US English)), and Force Account Code (unchecked). A red box highlights the Code, Feature, Telephone Number, and Line Group ID fields.

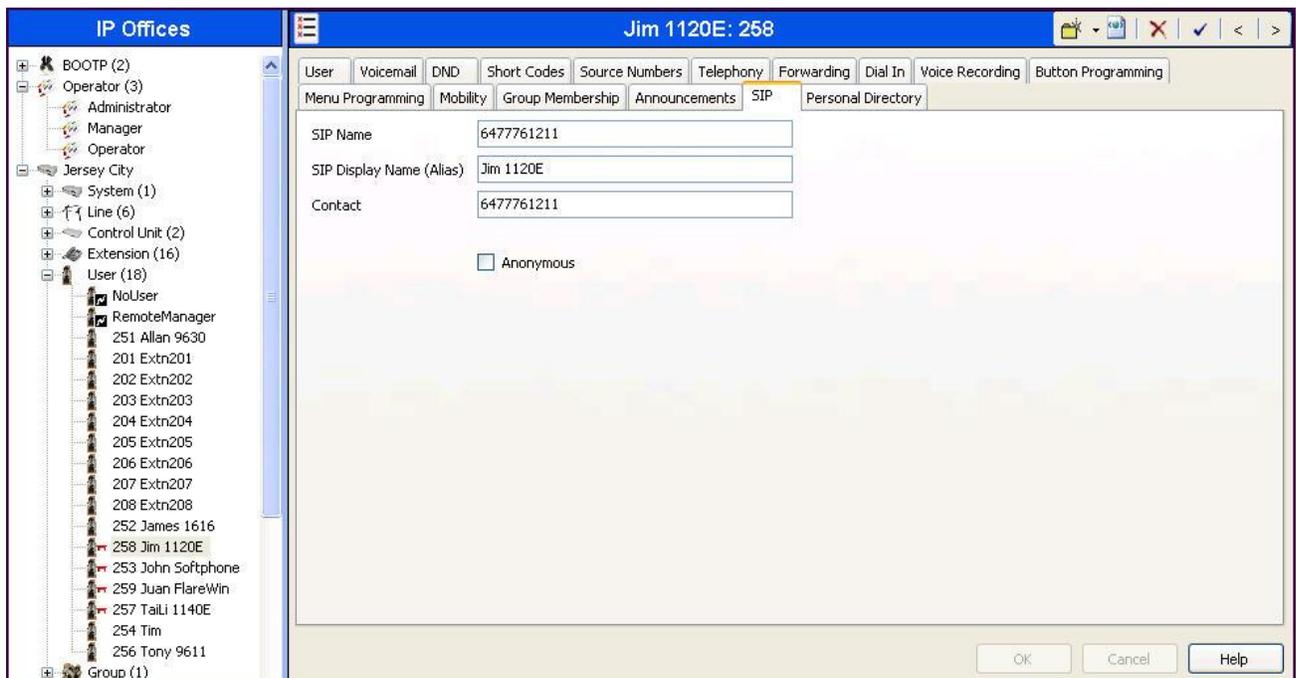
Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code **\*67N;** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter **W**, which means “withhold the outgoing calling line identification”. In the case of the compliance test, when a user dialed \*67 plus the number, Avaya IP Office would include the user’s telephone number (DID number assigned to the user) in the **P-Asserted-Identity** (PAI) header and would include the **Privacy: id** header in the outbound INVITE message. MTS Allstream would allow the call due to the presence of a valid DID in the PAI header, but would prevent presentation of the caller id to the called PSTN destination.

Short Code	
Code	*67N;
Feature	Dial
Telephone Number	WN"@192.168.2.12"
Line Group ID	17
Locale	
Force Account Code	<input type="checkbox"/>

## 5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Lines defined in **Section 5.4.2**. To configure these settings, first navigate to **User**→**Name** in the Navigation Pane, where **Name** is the name of the user to be modified. In the example below, the name of the user is **Jim 1120E**. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by MTS Allstream. The **SIP Display Name (Alias)** can optionally be configured with a descriptive text string. The value entered for the **Contact** field will be used in the Contact header for outgoing SIP INVITE to the service provider. The value entered for the **SIP Name** is used as the user part of the SIP URI in the From header for outgoing SIP trunk calls.

If outbound calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network (or alternatively use the \*67 short code as defined in **Section 5.5**).



## 5.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, right-click **Incoming Call Route** in the Navigation Pane and select **New** (not shown). 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 Group** of the SIP Line defined in **Section 5.4.2**.
- Set the **Incoming Number** to the incoming DID number on which this route should match. Matching is right to left.

The screenshot shows the configuration window for an Incoming Call Route. The left pane shows a tree view with 'Incoming Call Route (19)' expanded, and '17 6477761211' selected. The main pane shows the 'Standard' tab with the following fields:

Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	6477761211
Incoming Sub Address	
Incoming CLI	
Locale	United States (US English)
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to 6477761211 on Incoming Group 17 are to be routed to the user “Jim 1120E” at extension 258.

The screenshot shows the 'Destinations' tab of the configuration window. It contains a table with the following data:

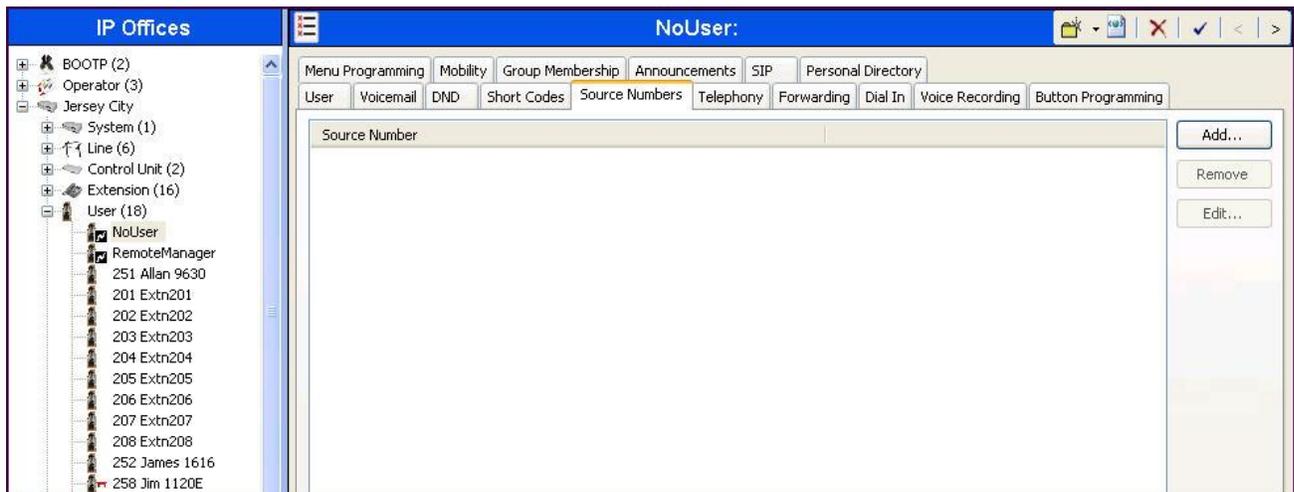
TimeProfile	Destination	Fallback Extension
Default Value	258 Jim 1120E	

## 5.8. SIP Options

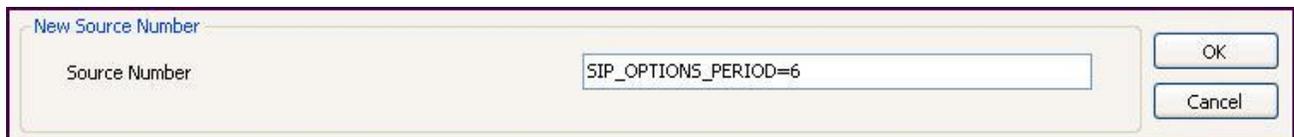
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. By default, Avaya IP Office Release 9.0 sends out OPTIONS every 300 seconds. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set 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:

- To use the default value, set **Binding Refresh Time** to 0 or 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP\_OPTIONS\_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time** setting.
- To establish a period greater than 300 seconds, a **SIP\_OPTIONS\_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP\_OPTIONS\_PERIOD** settings.

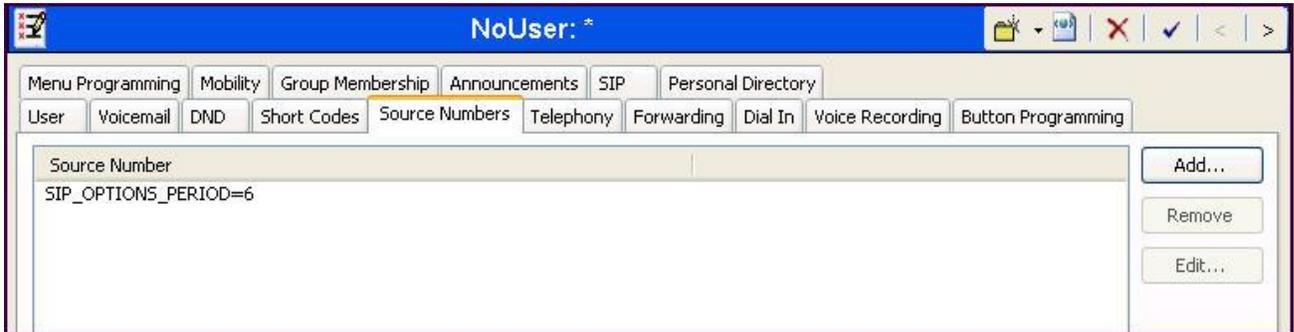
To configure the **SIP\_OPTIONS\_PERIOD** parameter, navigate to **User** → **noUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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



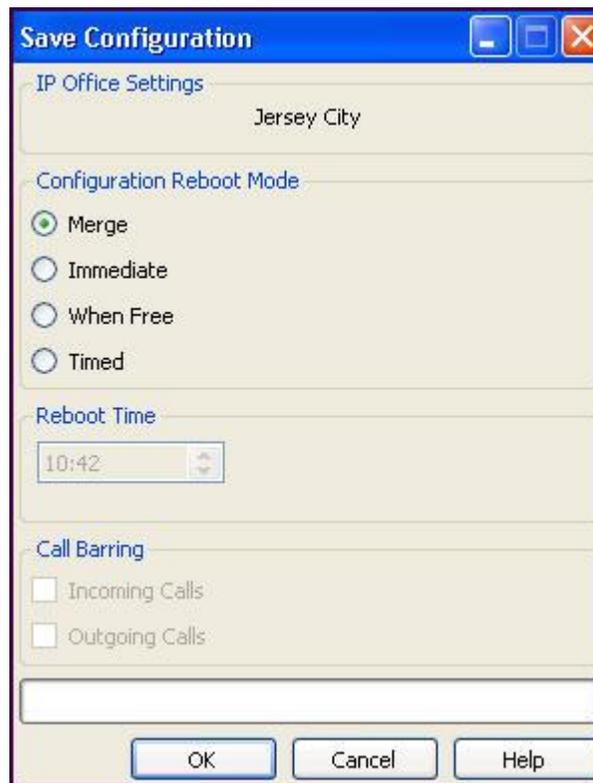
The **SIP\_OPTIONS\_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an **OPTIONS** period of 60 seconds was desired. The **Binding Refresh Time** was set to **60** seconds in **Section 5.2**. There was no need to define **SIP\_OPTIONS\_PERIOD**.



## 5.9. 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.

The following will appear, with either **Merge** or **Immediate** selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** to proceed.



## 6. MTS Allstream SIP Trunking Configuration

MTS Allstream is responsible for the configuration of its SIP Trunking service. The customer will need to provide the IP address used to reach the Avaya IP Office at the business site. MTS Allstream will provide the customer the necessary information to configure the Avaya IP Office SIP connection to MTS Allstream including:

- Network edge IP addresses of the MTS Allstream SIP Trunking service.
- Transport and port for the MTS Allstream SIP Trunking connection to the Avaya IP Office at the business site.
- DID numbers to assign to users at the business site.
- Supported codecs and their preference order.

## 7. Verification Steps

This section provides verification steps that may be performed in the field to verify that the solution is configured properly

### 7.1. System Status

Use the Avaya IP Office System Status application to verify the SIP Line channels state and to check alarms:

- Launch the application from **Start → Programs → IP Office → System Status** on the Avaya IP Office Manager PC. Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is *Idle* for channels where no active calls are currently in session; the state should be *Connected* for channels engaged in active calls.

**AVAYA IP Office System Status**

Help Snapshot LogOff Exit About

**System**

- Alarms (2)
- Extensions (12)
- Trunks (6)
  - Lines: 1 - 4
  - Line: 17
  - Line: 18
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

**SIP Trunk Summary**

Peer Domain Name: 192.168.96.231  
 Resolved Address: 192.168.2.12  
 Line Number: 17  
 Number of Administered Channels: 12  
 Number of Channels in Use: 2  
 Administered Compression: G729 A, G711 Mu  
 Silence Suppression: Off  
 Layer 4 Protocol: UDP  
 SIP Trunk Channel Licenses: Unlimited 1%  
 SIP Trunk Channel Licenses in Use: 2  
 SIP Device Features: UPDATE (Incoming and Outgoing)

Channel Number	URI G...	Call Ref	Current State	Time in State	R.. Codec	C.. C..	Other Party on Call	Direction of Call	Round Trip D...	Receive Jitter	Receive Packet...	Transmit Jitter	Transmit Packet...
1	1	10	Connected	00:05:30	...	G729 A	...	Extrn 256, Tony 9611	Incoming	20ms		1.1ms	0%
2	0	11	Connected	00:05:03	...	G729 A	...	Extrn 258, Jim 1120E	Outgoing	14ms		0.1ms	0%
3			Idle	22:26:31									
4			Idle	22:26:31									
5			Idle	22:26:31									
6			Idle	22:26:31									
7			Idle	22:26:31									
8			Idle	22:26:31									
9			Idle	22:26:31									
10			Idle	22:26:31									

Trace Trace All Pause Ping Call Details Print... Save As...

- Select the **Alarms** tab and verify that no alarms are active on the SIP line.

Alarms for Line: 17 SIP 192.168.96.231		
Last Date Of Error	Occurrences	Error Description

## 7.2. Monitor

The Monitor application can be used to monitor and troubleshoot Avaya IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor** on the Avaya IP Office Manager PC. The application allows the monitored information to be customized. To customize, select **Filters → Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, **Standard** SIP Events and the **SIP Rx** and **SIP Tx** boxes are checked.

The screenshot shows the 'All Settings' dialog box with the 'SIP' tab selected. The dialog is organized into several sections:

- Events:** Contains three checked options:  **Sip** (with a dropdown menu set to 'Standard'),  **STUN**, and  **SIP Dect**.
- Packets:** Contains eight unchecked options:  SIP Reg/Opt Rx,  SIP Reg/Opt Tx,  SIP Call Rx,  SIP Call Tx,  SIP Misc Rx,  SIP Misc Tx,  Cm Notify Rx, and  Cm Notify Tx.
- Bottom Section:** Contains two checked options:  **Sip Rx** and  **Sip Tx**. To the right is an 'IP Filter (nnn.nnn.nnn.nnn)' label and an empty text input field.

At the bottom of the dialog, there are two rows of buttons: **Default All**, **Clear All**, **Tab Clear All**, **Tab Set All**, **OK**, **Cancel** in the first row; and **Save File**, **Load File**, **Load Partial File**, **Select File** in the second row.

## 8. Conclusion

The MTS Allstream SIP Trunking service passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office R9.0 and MTS Allstream SIP Trunking as shown in **Figure 1**. Test results and observations are noted in **Section 2.2**.

## 9. Additional References

- [1] *IP Office Documentation Library*, Release 9.0, Documentation number 15-604278 Issue 1, September 2013
- [2] *IP Office 9.0 Product Description*, Documentation number 15-601041 Issue 27.01.0, September 2013.
- [3] *Avaya IP Office 9.0 Installing IP500/IP500 V2*, Document number 15-601042 Issue 28g, October 2013.
- [4] *Avaya IP Office 9.0 Administering Voicemail Pro*, Document number 15-601063 Issue 9.01.0, September 2013.
- [5] *Avaya IP Office Manager Release 9.0*, Document number 15-601011 Issue 9.01, September 2013.
- [6] *Avaya IP Office 9.0 Using System Status*, Document number 15-601758 Issue 09c, August 2013.
- [7] *Avaya IP Office 9.0 Using IP Office System Monitor*, Document Number 15-601019, Issue 05c, August 2013.
- [8] *Avaya IP Office 9.0 H.323 Telephone Installation*, Document Number 15-601046, Issue 18b, August 2013.
- [9] *Avaya IP Office 9.0 SIP Extension Installation*, Issue 3c, August 2013.

Additional IP Office documentation can be found at:

<http://marketingtools.avaya.com/knowledgebase/>.

Product documentation for the MTS Allstream SIP Trunking is available from MTS Allstream. See **Section 2.3** on how to contact MTS Allstream.

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