Avaya Solution & Interoperability Test Lab

Application Notes for Configuring MTS Allstream SIP Trunking with Avaya IP Office - Issue 1.0

Abstract

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

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 the service provider MTS Allstream and Avaya IP Office.

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.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the MTS Allstream SIP Trunking service via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Avaya IP Office and various Avaya endpoints.

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries
- 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 over 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 over the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients
  Avaya IP Office supports two soft clients: Avaya IP Office Phone Manager and Avaya IP Office Video Softphone. Avaya IP Office Phone Manager supports two modes (PC softphone and telecommuter). Both clients in each supported mode were tested.
- Various call types including: local, long distance, international, outbound toll-free, operator services and directory assistance
- Codec G.711MU and G.729A
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Response to incomplete call attempts and trunk errors
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning
- G.711 pass-through fax

Items not supported or not tested include the following:

- MTS Allstream SIP Trunking was not configured to send SIP OPTIONS messages during the compliance test but will respond to the OPTIONS messages sent by Avaya IP Office.
- Inbound toll-free and emergency calls (911) are supported but were not tested as part of the compliance test.

- Local outbound calling using 7 digit dialing is not supported. These calls require dialing 10 digits. Inbound local calls can be configured for 7 digits but this was not tested.
- T.38 fax is not supported.
- The SIP REFER method is not supported for network redirection.

2.2. Test Results

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

- **No error indication for outbound call with no matching codec –** If Avaya IP Office is misconfigured so that no supported codec is offered to MTS Allstream on an outbound call, MTS Allstream will return a “488 Not Acceptable Here” response. Avaya IP Office will disconnect the call without providing any error indication to the original enterprise caller.

2.3. Support

For technical support on the MTS Allstream SIP Trunking service, contact MTS Allstream Customer Care by calling 866-282-0111 or by sending email to ABC3@mtsallstream.com.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. Selecting the **Support Contact Options** link followed by **Maintenance Support** provides the worldwide support directory for Avaya Global Services. Specific numbers are provided for both customers and partners based on the specific type of support or consultation services needed. Some services may require specific Avaya service support agreements. 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. The test configuration shows an enterprise site connected to MTS Allstream SIP Trunking.

Located at the enterprise site is an Avaya IP Office 500. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public network. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 5600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya IP Office Phone Manager, an Avaya IP Office Softphone, an Avaya 5420 Digital Telephone, an Avaya 6210 Analog Telephone and a fax machine. The site also has a Windows 2003 Server running Avaya Voicemail Pro for voicemail and running Avaya IP Office Manager to configure the Avaya IP Office.

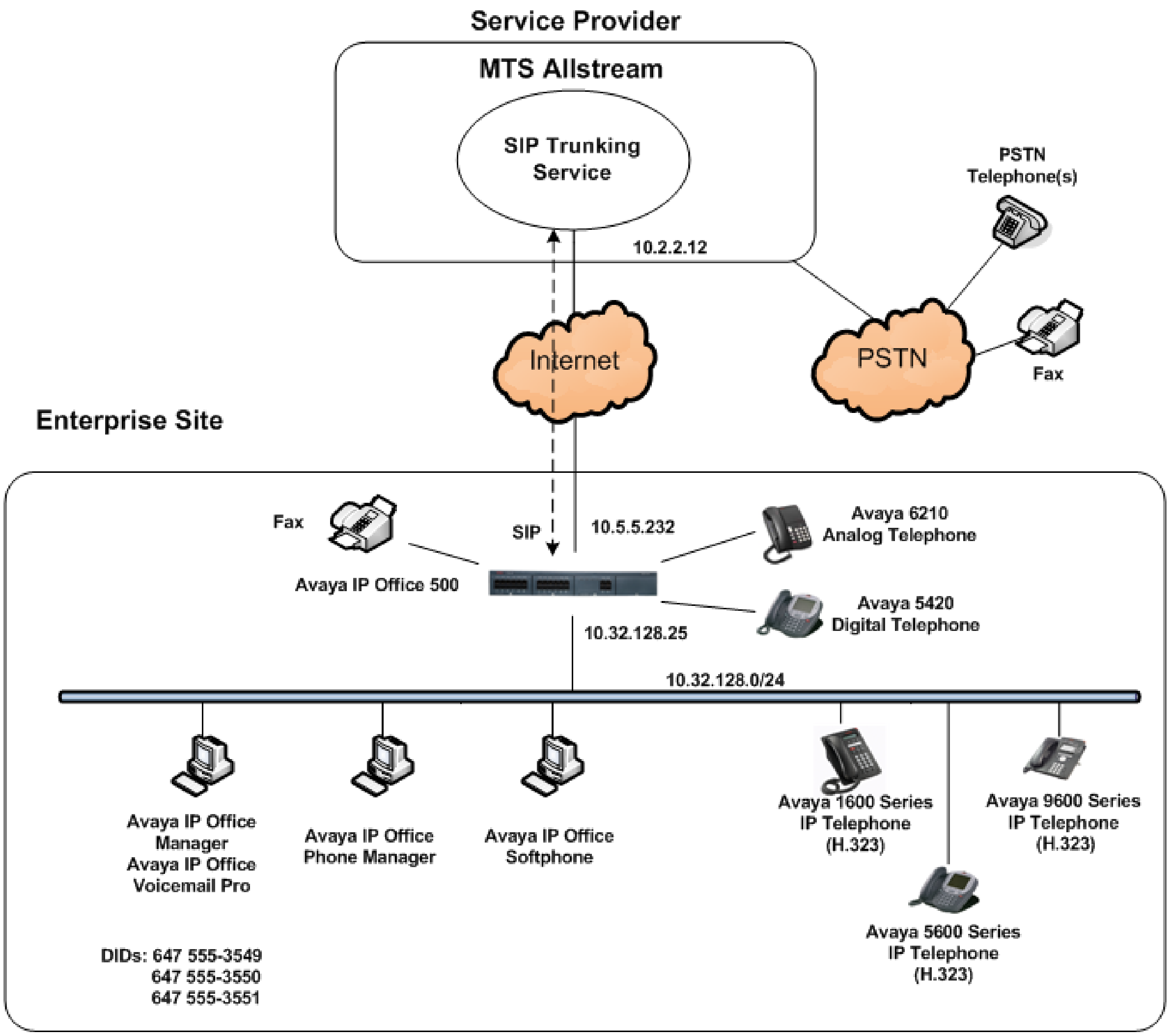


Figure 1: Test Configuration

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 numbers that can not be routed by the PSTN.

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 unaltered to MTS Allstream. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits. 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. It was configured to send 10 digits in the From header. For inbound calls, MTS Allstream sent 10 digits in the Request URI and the To header of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the 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 the 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:

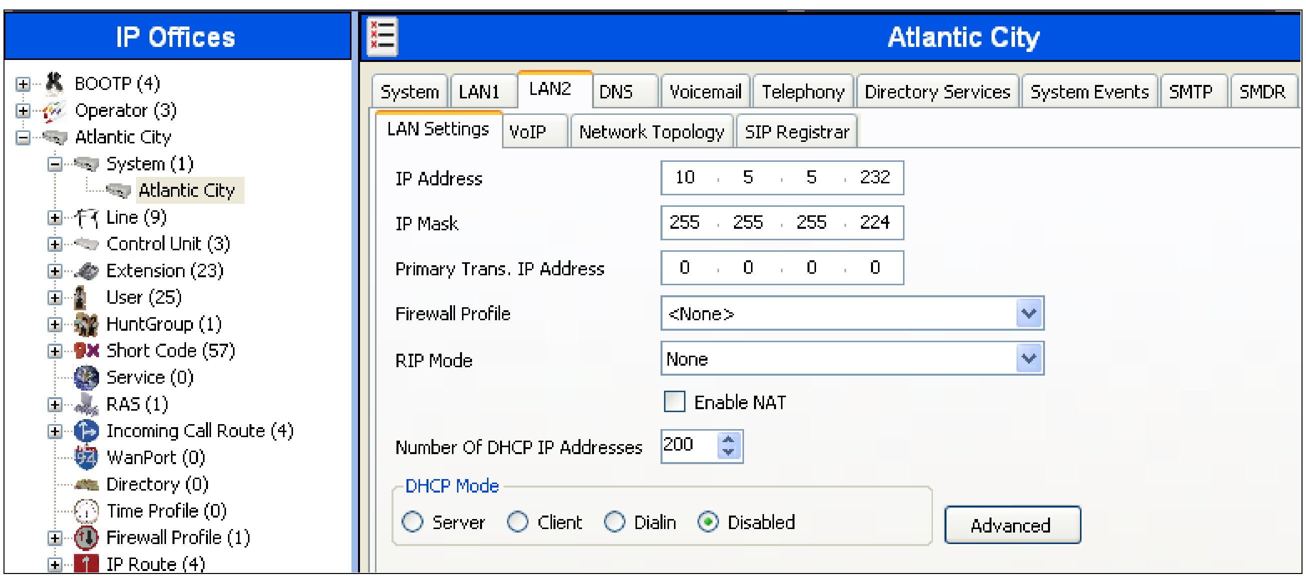
Avaya Telephony Components	
Equipment	**Release**
Avaya IP Office 500	8.0 (16)
Avaya IP Office Manager	10.0 (16)
Avaya Voicemail Pro	8.0 (Build 8.0.8.29)
Avaya 1608SW IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.300B
Avaya 5620 IP Telephone (H.323)	2.9.1
Avaya 9640SW IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.102S
Avaya IP Office Phone Manager	4.2.36
Avaya IP Office Softphone	3.2.3.15 (64595)
Avaya 5420 Digital Telephone	N/A
Avaya 6210 Analog Telephone	N/A
MTS Allstream Components	
Equipment	**Release**
Genband S3 Session Border Controller	5.2.2.12
Nortel CS2K	CVM13

5. Configure Avaya IP Office

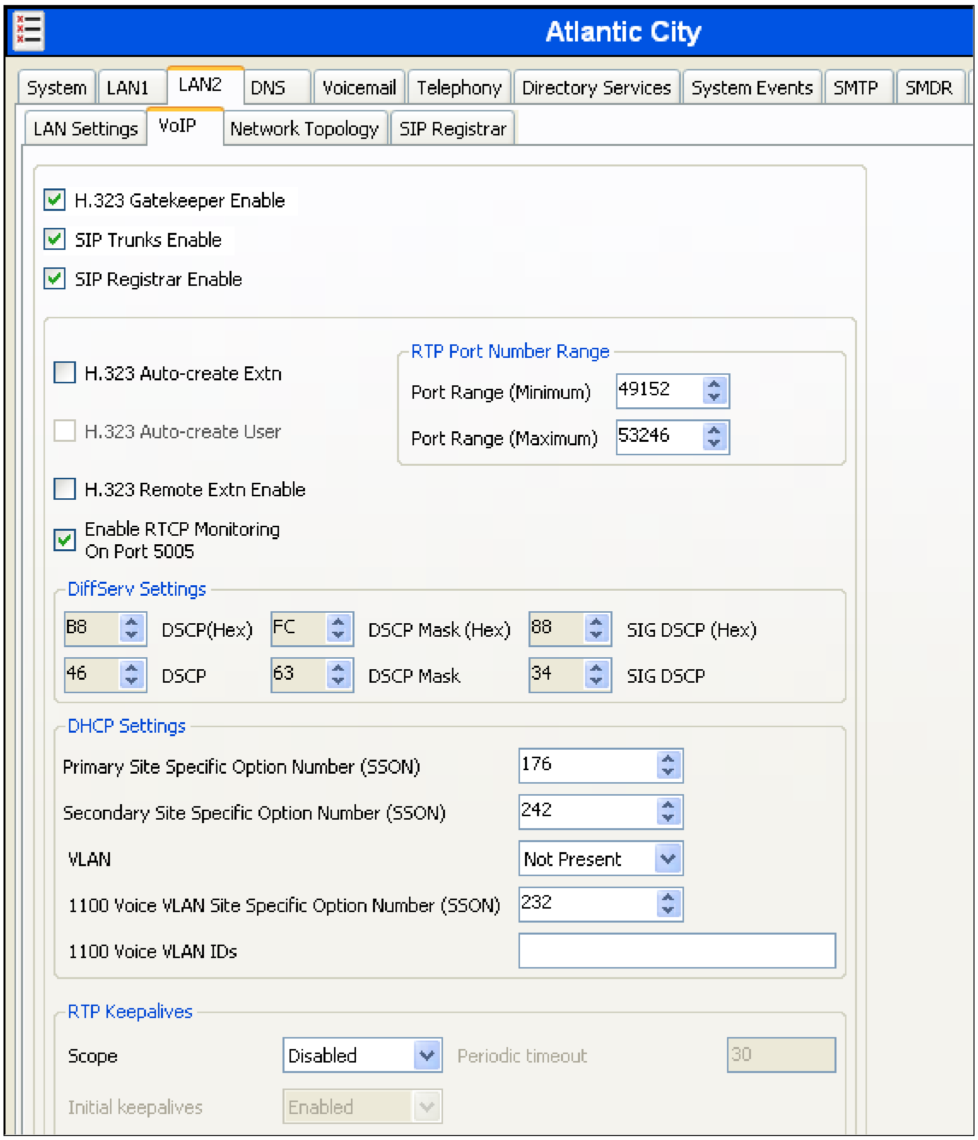
This section describes the Avaya IP Office configuration to support connectivity to MTS Allstream SIP Trunking. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya 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 Avaya 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 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.

5.1. LAN2 Settings

In the sample configuration, ***Atlantic City*** was used as the system name and the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office. To access the LAN2 settings, first navigate to **Atlantic City → System → Atlantic 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. All other parameters should be set according to customer requirements.

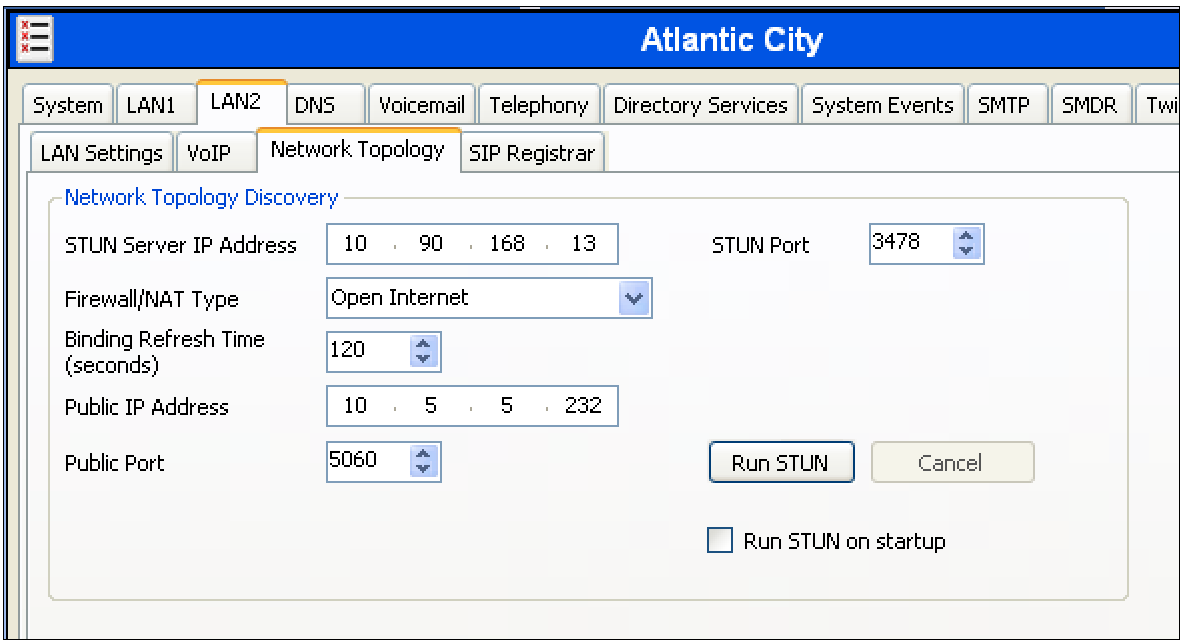


On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP 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.



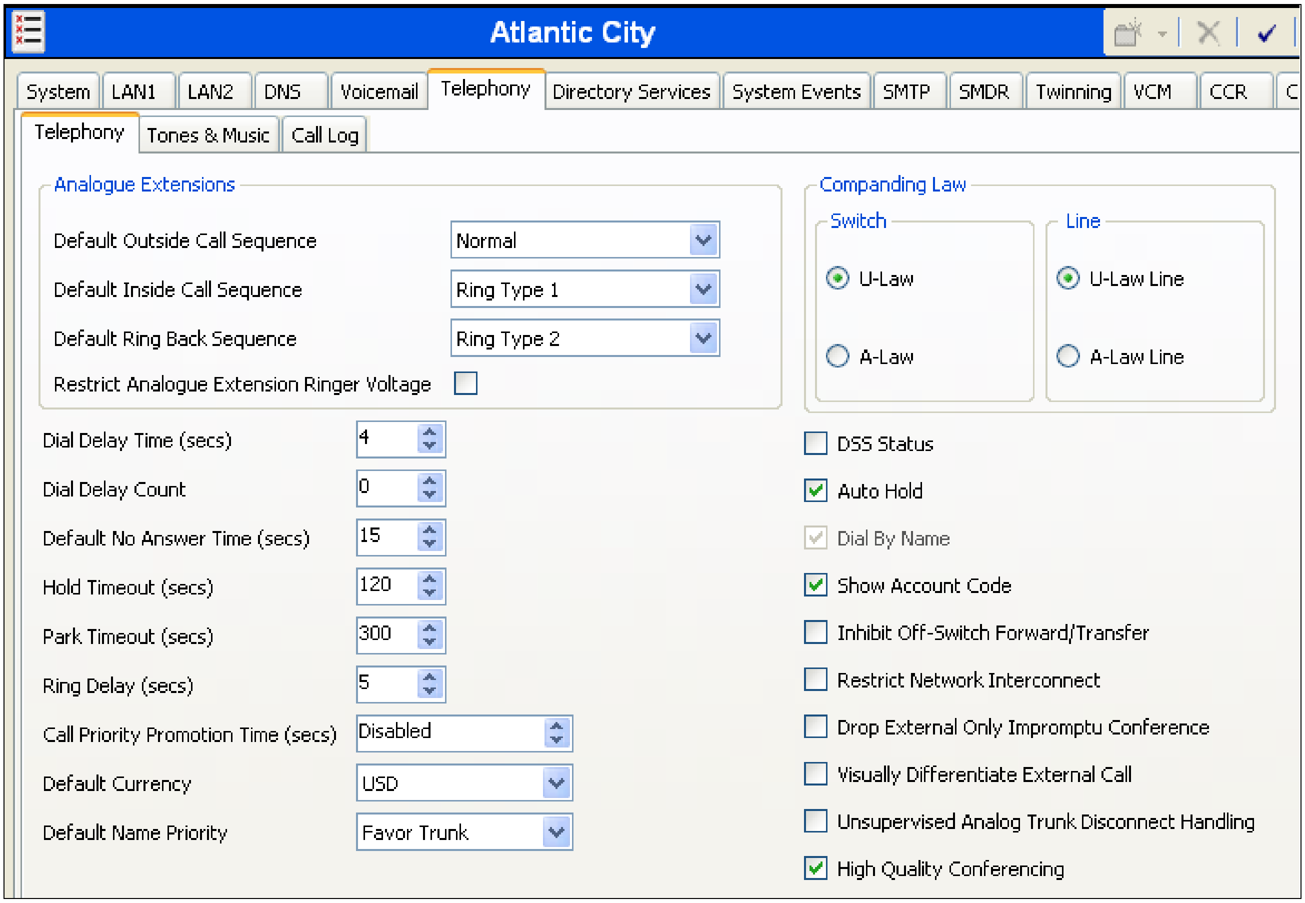
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- 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***. Since this parameter is set to ***Open Internet***, then the **STUN Server IP Address** is not used.
- Set **Binding Refresh Time (seconds)** to ***120***. 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 5.9** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set the **Public Port** to ***5060***.
- All other parameters should be set according to customer requirements.



5.2. System Telephony Settings

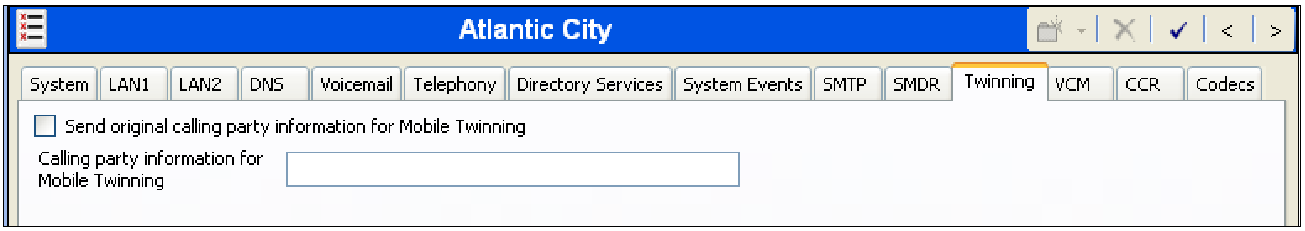
Navigate to the **Telephony → Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For the compliance test, ***ULAW*** was used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk.



5.3. Twinning Calling Party Settings

Navigate to the **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning and Call Forwarding to be controlled by the setting on the SIP Line (**Section 5.4**).

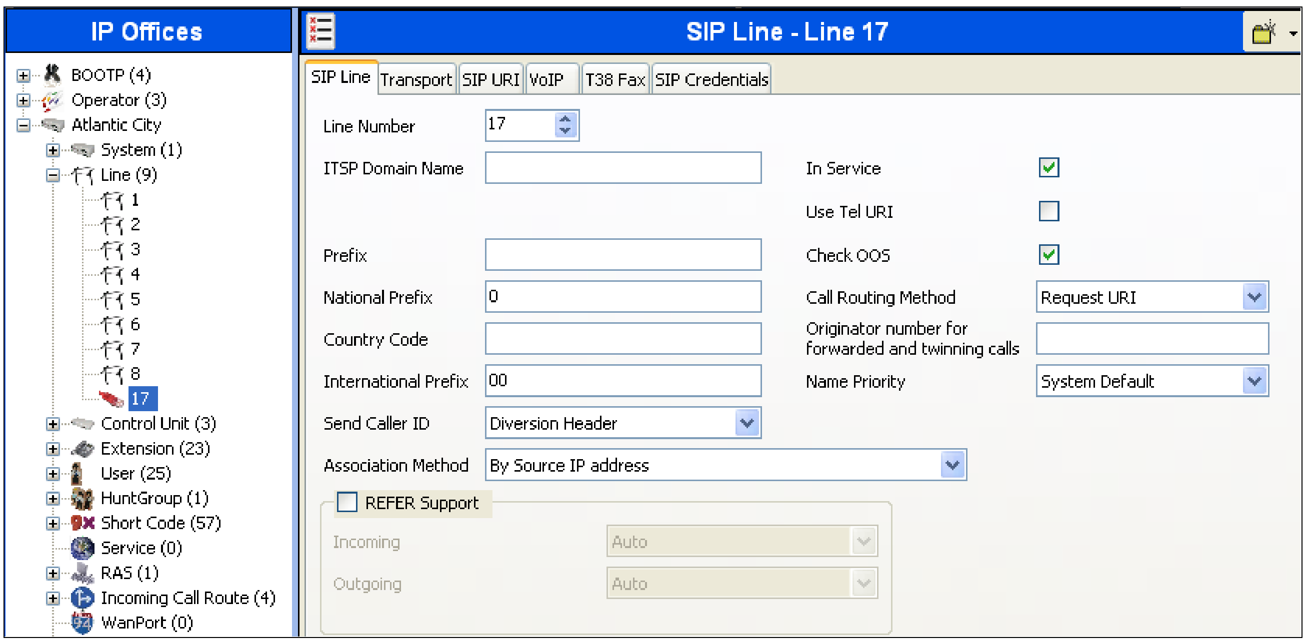
Click the **OK** Button at the bottom of the page (not shown).



5.4. Administer SIP Line

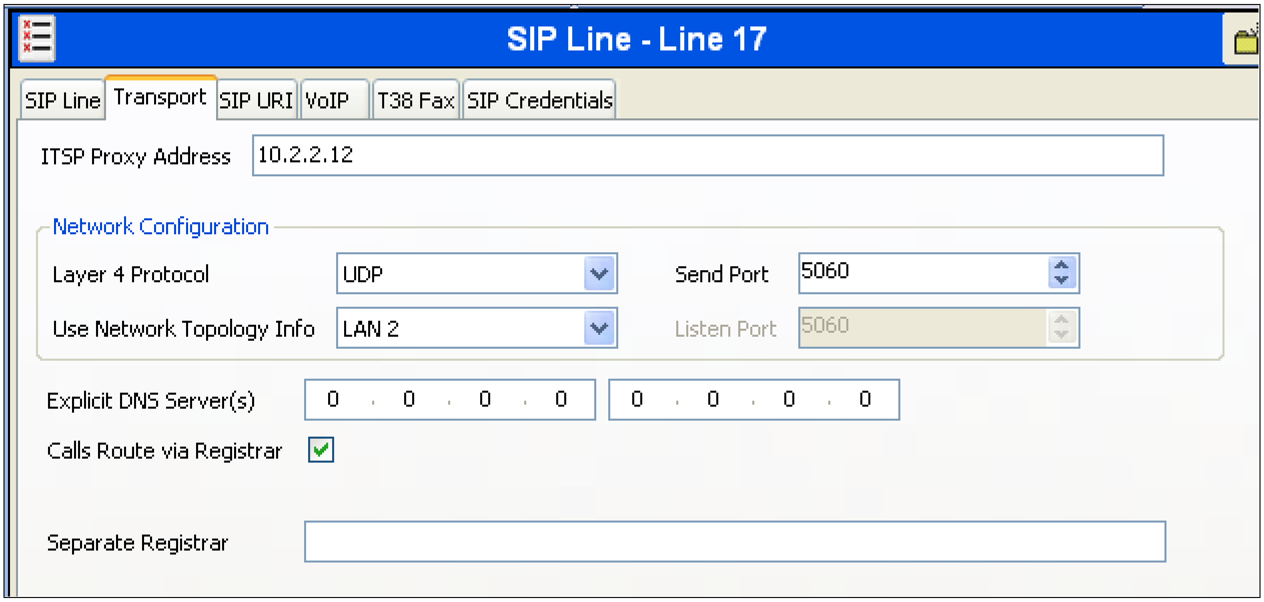
A SIP line is needed to establish the SIP connection between Avaya IP Office and MTS Allstream SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New → SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure the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

- Check the **In Service** box.
- Check the **Check OOS** box.
- Set the **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***. For forwarded or twinning calls, this setting results in the original calling party appearing in the SIP From header and the forwarding/twinning party in the Diversion header.
- Uncheck the **REFER Support** box since MTS Allstream does not support REFER.
- Default values may be used for all other parameters.



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

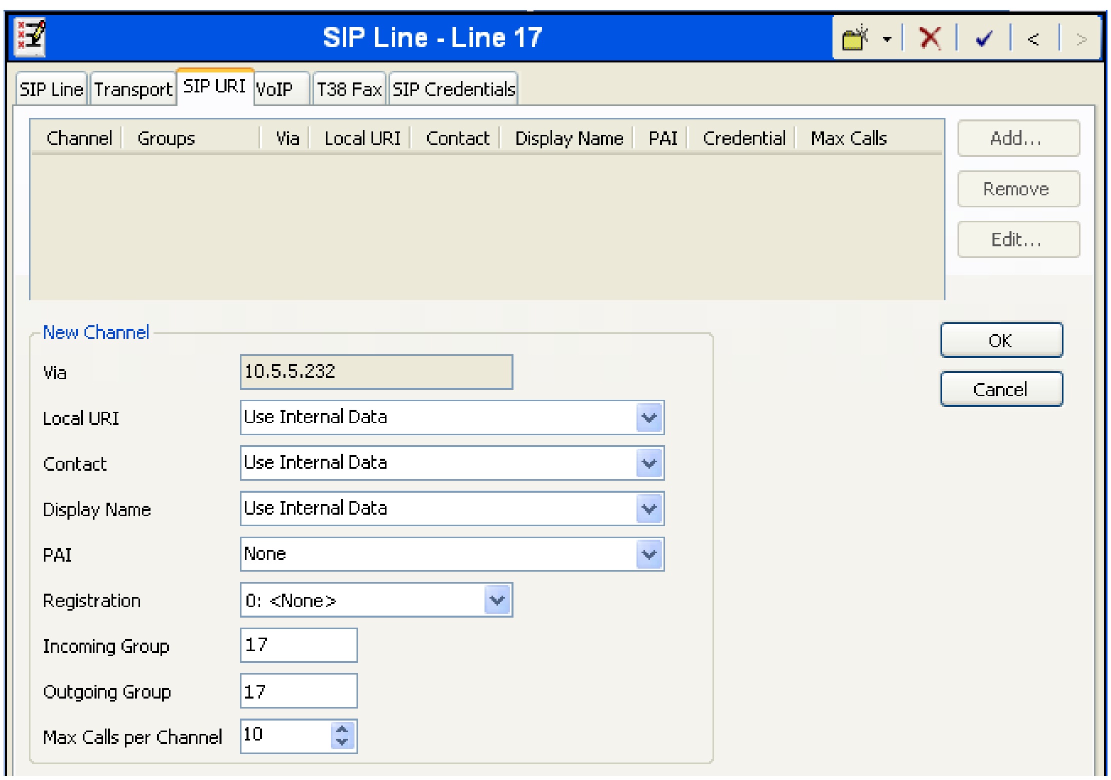
- Set the **ITSP Proxy Address** to the IP address of the MTS Allstream SIP Proxy provided by MTS Allstream.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to ***LAN2*** as configured in **Section 5.1**.
- Set the **Send Port** to ***5060***.
- Default values may be used for all other parameters.



Select the **SIP URI** tab, to create a SIP URI entry. A SIP URI entry must be created to match 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 that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

- 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 5.6**.
- For the **Registration** field, select **0:<None>** from the pull-down menu.
- 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 ***17*** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Default values may be used for all other parameters.

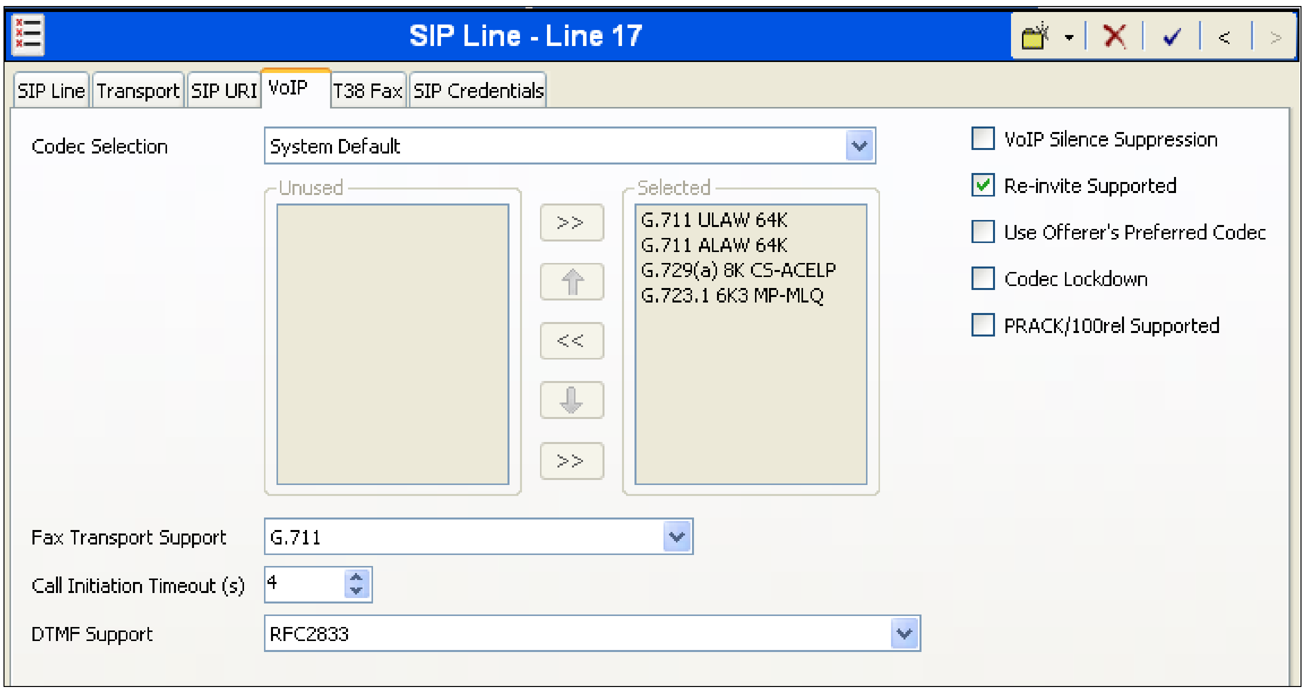
Click **OK**.



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

- Set the **Codec Selection** to ***System Default***. The default codec list and their order are shown in the **Selected** box.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Set **Fax Transport Support** to ***G.711*** to use G.711 pass-through fax.
- Set the **DTMF Support** field to ***RFC2833***. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters.

Click the **OK** button at the bottom of the page (not shown).

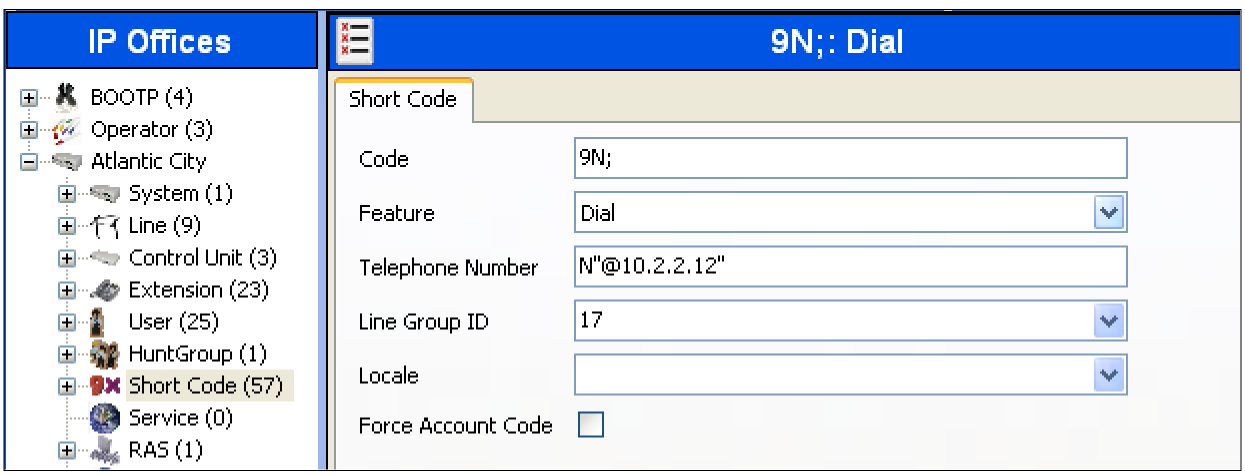


5.5. Short Code

Define a short code to route outbound traffic 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. 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"@10.2.2.12"***. 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 MTS Allstream SIP IP address follows the @ sign in the above expression.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing the outbound call.
- Default values may be used for all other parameters.

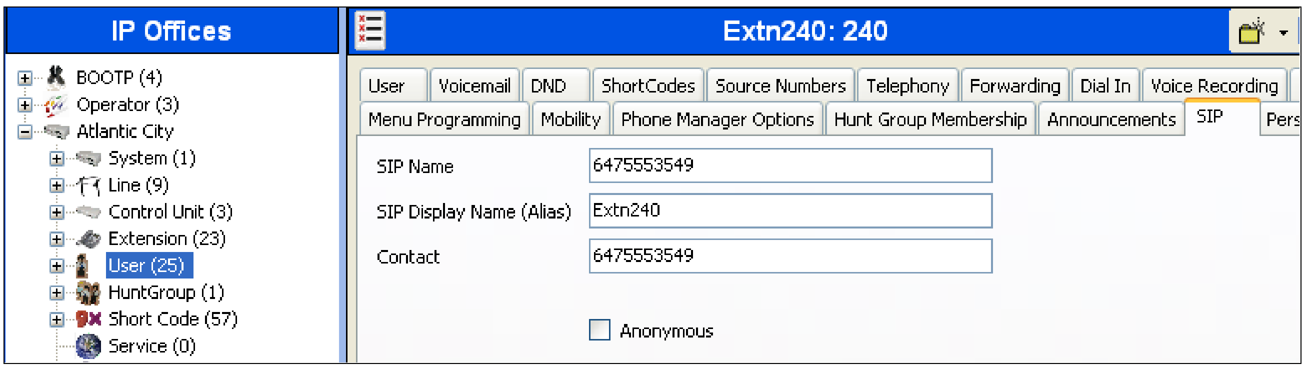
Click the **OK** button (not shown).



5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. 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 ***Extn240***. 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 from MTS Allstream. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. The value entered for the **Contact** field will be used in the SIP INVITE for outgoing SIP trunk calls. 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.

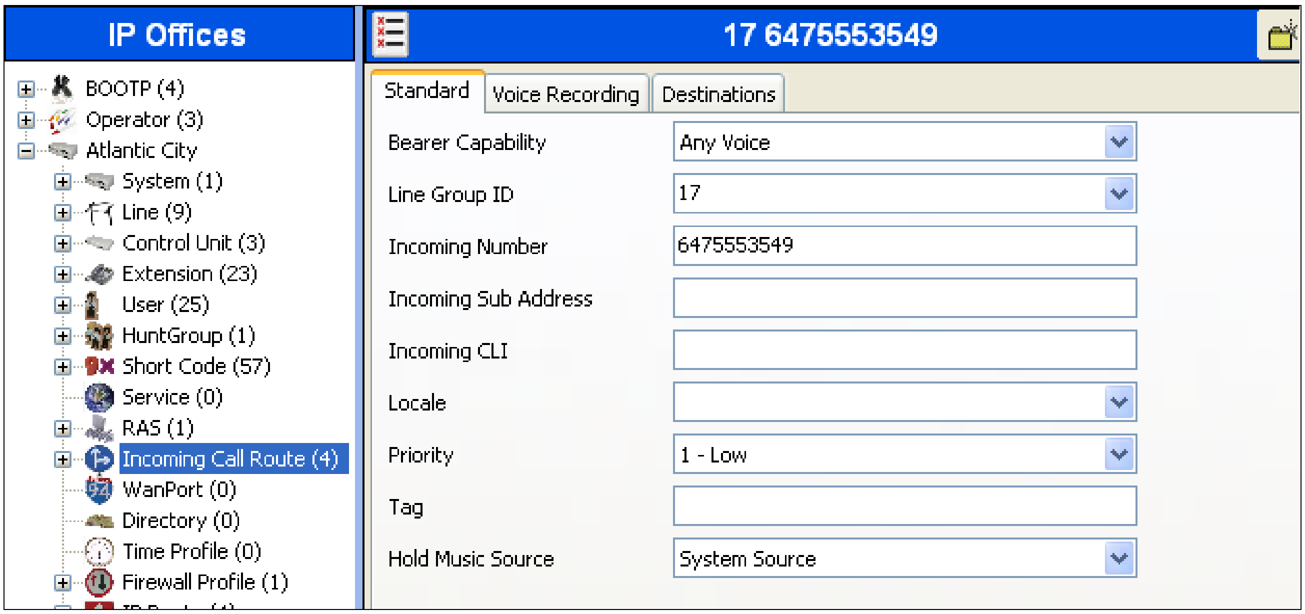
Click the **OK** button (not shown).



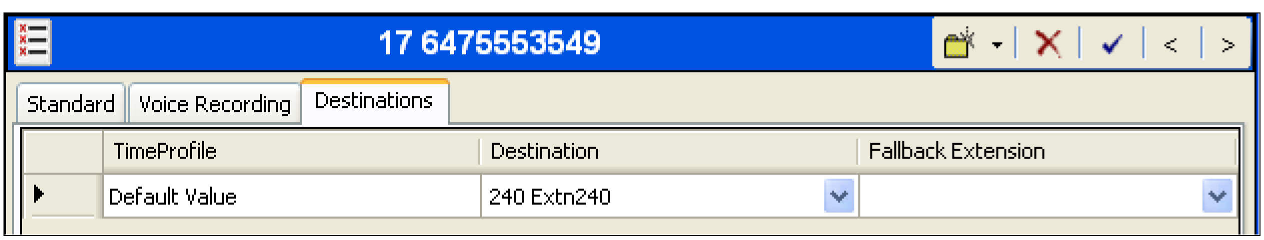
5.7. Incoming Call Route

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 Routes** 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 line group of the SIP line defined in **Section 5.4**.
- Set the **Incoming Number** to the incoming number on which this route should match. Matching is right to left.
- Default values can be used for all other fields.



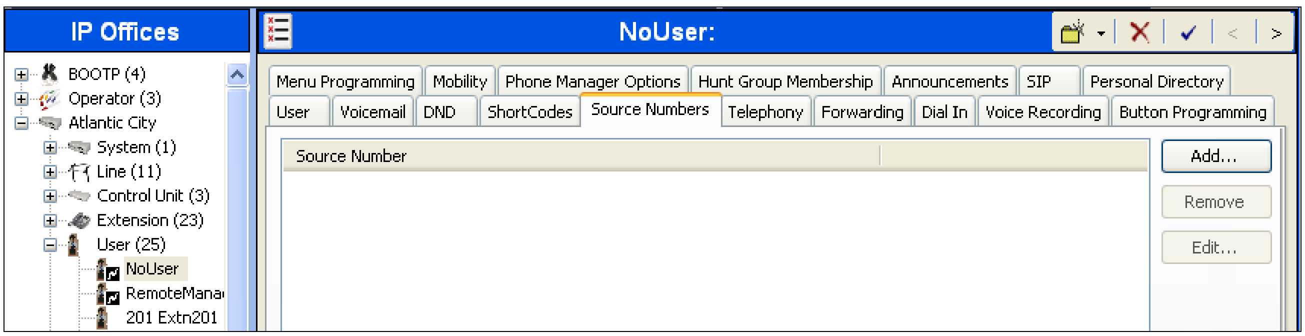
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 6475553549 on line 17 are routed to extension 240.



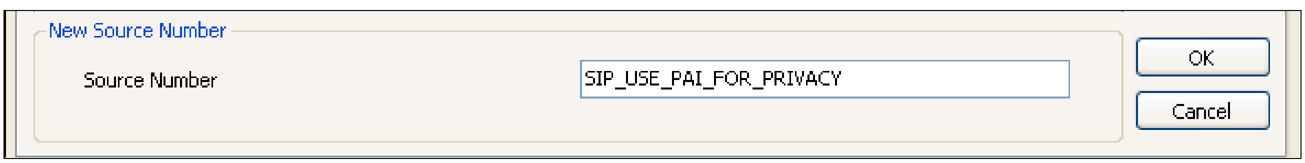
5.8. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “restricted” and “anonymous” respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. By default, Avaya IP Office will use PPI for privacy. MTS Allstream supports both PPI and PAI for purposes of privacy. For the compliance test, PPI was used for the purposes of privacy.

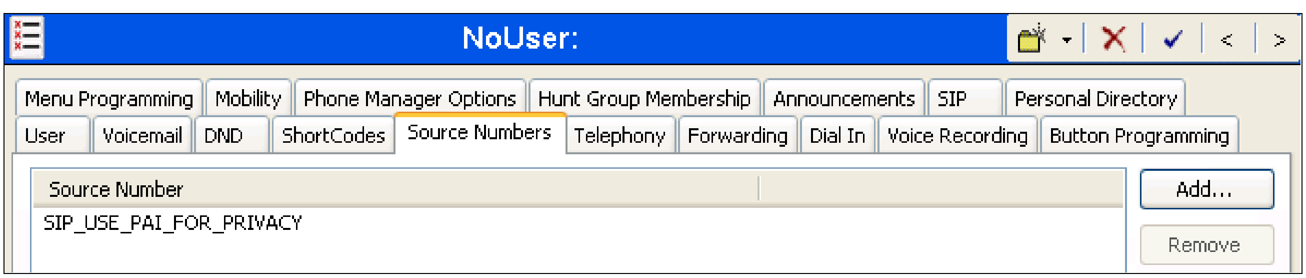
If Avaya IP Office was to be configured to use PAI for privacy calls, then perform the following. Navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter ***SIP_USE_PAI_FOR_PRIVACY***. Click **OK**.



The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.



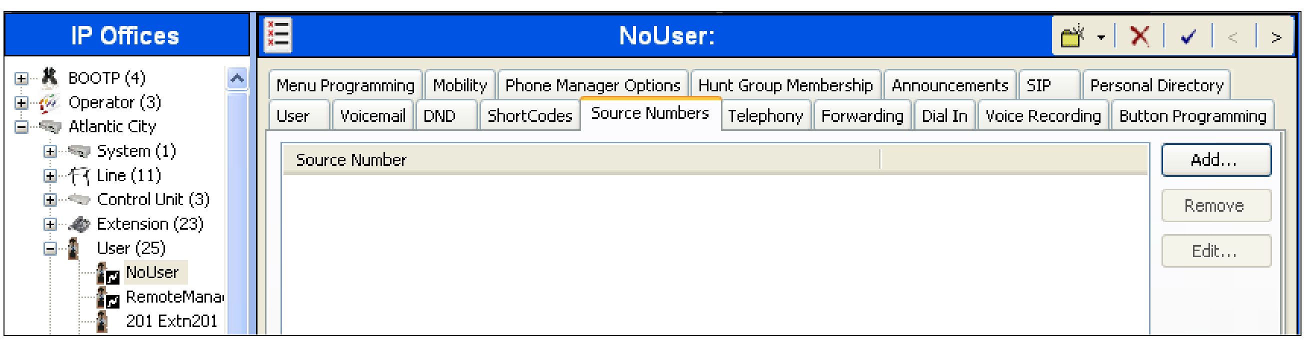
5.9. SIP Options

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 5.1** 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:

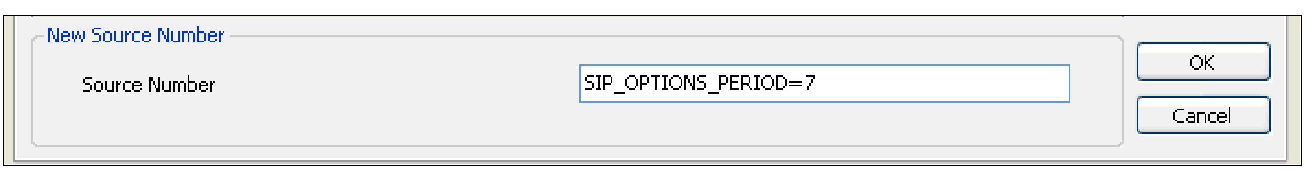
- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 300 seconds is used.
- To establish a period less than 300 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- 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 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

For the compliance test, an OPTIONS period of 2 minutes was desired. Thus, the **Binding Refresh Time** was set to ***120*** seconds (2 minutes) in **Section 5.1** and no **SIP_OPTIONS_PERIOD** parameter was defined.

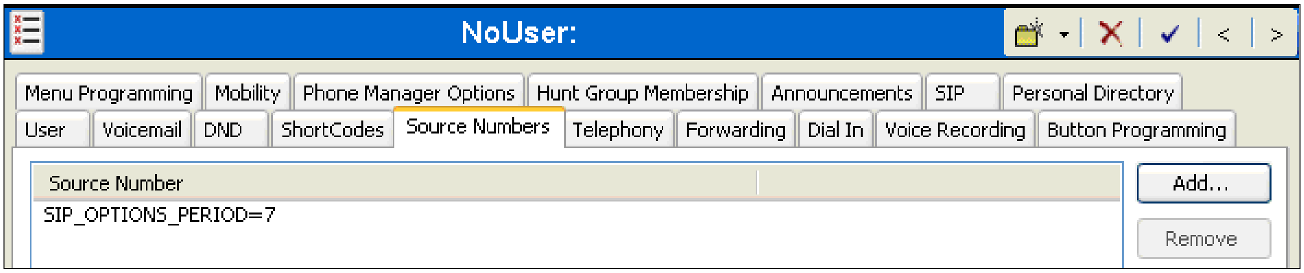
Alternatively, if an OPTIONS period greater than 300 seconds is desired then define the **SIP_OPTIONS_PERIOD** by doing the following. Navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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. 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. MTS Allstream SIP Trunking Configuration

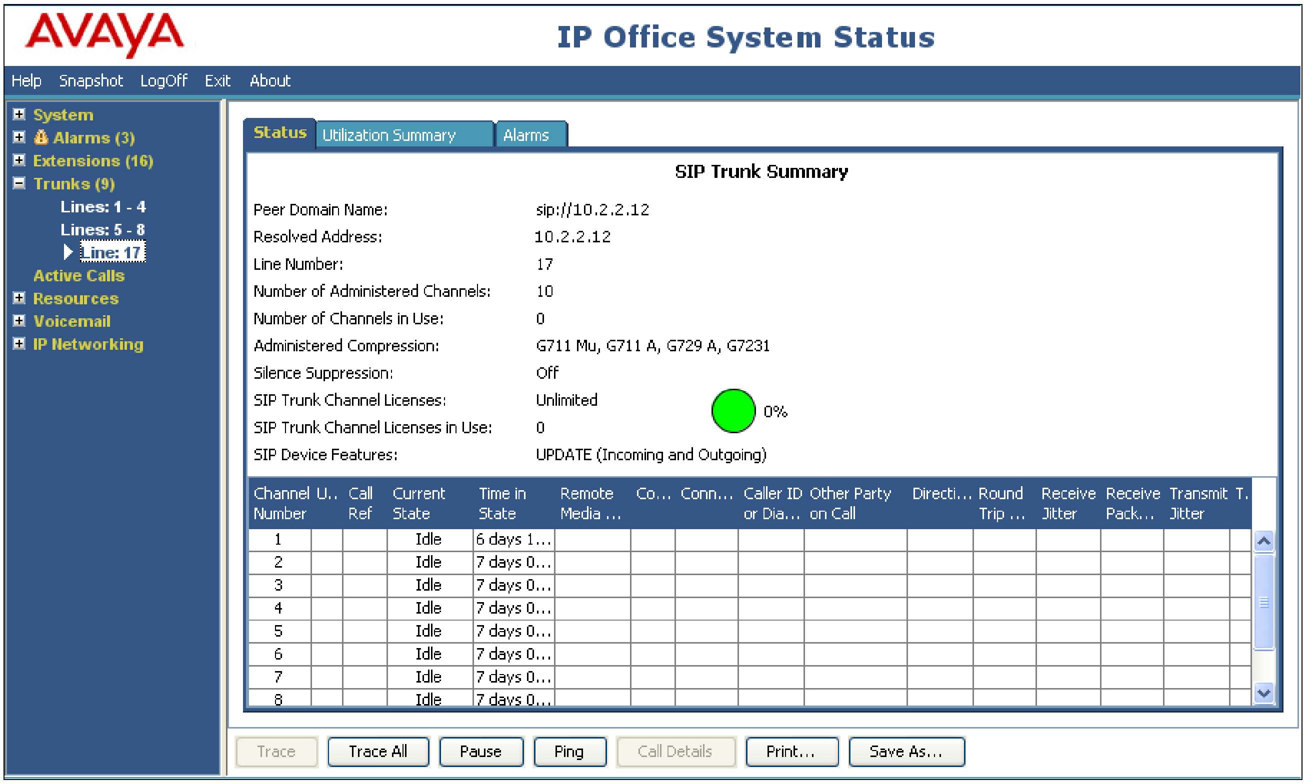
MTS Allstream is responsible for the configuration of MTS Allstream SIP Trunking. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. MTS Allstream will provide the customer the necessary information to configure the Avaya IP Office SIP connection to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

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

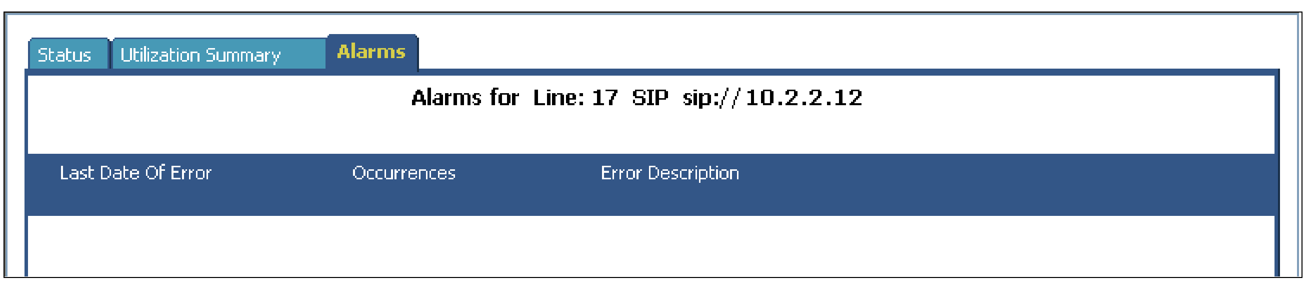
7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is ***Idle*** for each channel.



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



- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Verify that a phone connected to PSTN can successfully place a call to the Avaya IP Office with two-way audio.

8. Conclusion

MTS Allstream SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and MTS Allstream SIP Trunking as shown in **Figure 1**.

9. Additional References

[1] *IP Office Documentation CD*, December 2011
[2] *IP Office Installation Manual,* Document number 15-601042, December 2011.
[3] *IP Office Manager Manual,* Document number 15-601011, December 2011.
[4] *IP Office System Status Application,* Document number 15-601758, November 2011.

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for MTS Allstream SIP Trunking is available from MTS Allstream. See **Section 2.3** on how to contact MTS Allstream.

Appendix: SIP Line Template

Avaya IP Office Release 8.0 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Not all of the configuration information is included in the SIP Line Template. Therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using **Section 5.4** in these Application Notes as a reference.

The SIP Line Template created from the configuration as documented in these Application Notes is as follows:

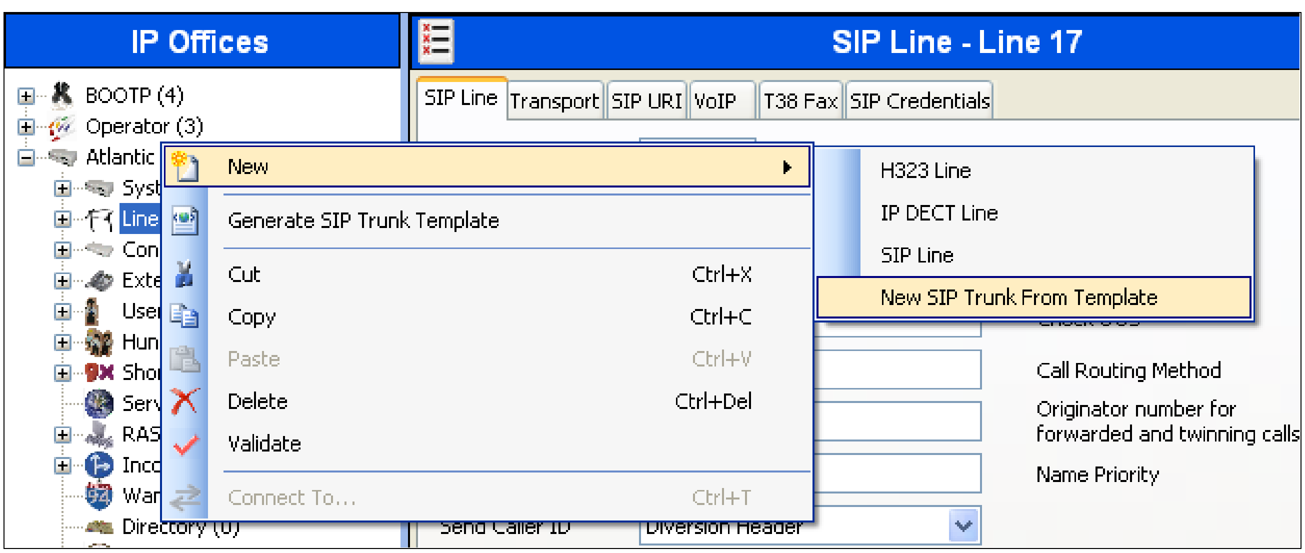

```
<?xml version="1.0" encoding="utf-8" ?>
- <Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20120119</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>MTS Allstream Trunk</DescriptiveName>
  <ITSPDomainName>10.2.2.12</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>false</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <ITSPProxy>10.2.2.12</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>false</UseAdvVoiceCodecPrefs>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_G711</FaxTransportSupport>
```

```xml
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
  <T38FaxVersion>3</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_14400</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>true</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>false</DisableT30ECM>
  <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
  <DisableT30MRCompression>false</DisableT30MRCompression>
  <NSFOverride>false</NSFOverride>
</Template>
```

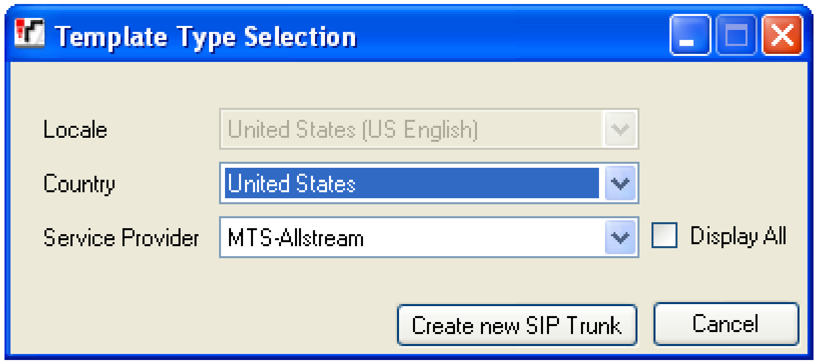

To import the above template into a new installation:

1. On the PC where IP Office Manager is installed, copy and paste the above template into a text document named **US_MTS-Allstream_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates).

2. Import the template into an IP Office installation by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New → New SIP Trunk From Template**:



3. Verify that ***United States*** is automatically populated for **Country** and ***MTS-Allstream*** is automatically populated for **Service Provider** in the resulting **Template Type Selection** screen as shown below. Click **Create new SIP Trunk** to finish the importing process.



Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.