**Avaya Solution & Interoperability Test Lab**

# Application Notes for Configuring Windstream with Avaya IP Office (7.0.5) - Issue 1.0

## Abstract


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

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

Windstream 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 service provider Windstream and Avaya IP Office.

Windstream SIP Trunking provides PSTN access via a SIP trunk between the enterprise and the Windstream 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 Windstream SIP Trunk 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 / Results

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

- Response to SIP OPTIONS queries
- Incoming PSTN calls to various Avaya phone types. Phone types included H.323, SIP, 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, SIP, 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 soft clients Avaya IP Office supports two soft clients: Avaya IP Office Phone Manager and Avaya IP Office Softphone. Only Avaya IP Office Softphone was tested.
- Various call types including: local, long distance, international, outbound toll-free and
- Directory assistance
- Codec G.711MU
- Caller ID presentation and Caller ID restriction
- 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 Faxing inbound and outbound
- SIP REFER on transfer and forward

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls (911) are supported but were not tested as part of the compliance test.
- Operator services are supported but were not tested.

- Codec negotiation of multiple codecs between Windstream and Avaya was not tested since Windstream currently supports only one codec (G.711MU) for SIP Trunking.

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

- **SIP OPTIONS** message response**:** IP Office sends SIP OPTIONS messages to the Windstream SIP gateway and Windstream returns a **501 Not Implemented**. A **200 OK** is the usual response but the IP Office only requires a response, not a specific message. However, **Check OOS** on the **Line →SIP Line** must be unchecked for IP Office to consider the SIP Line in-service. See **Section 5.5** for more details.
- **Codec mismatch on Outbound calls**: When incompatible codecs are configured, outbound calls succeed but caller gets dead air and then the call drops and phone displays "**INCOMPATIBLE**". Inbound calls do not go through and caller hears an appropriate rejection message.
- **RFC2833 Not Supported**: Only Inband DTMF is supported.
- **T.38 FAX Not Supported**: Only G.711 Fax is supported.
- **Outbound Call Number Restriction Not Supported:** On Outbound calls from the Avaya IP Office when Called-ID is restricted, Windstream does not allow that call to go through. Inbound calls with restricted caller-id complete and display "Anonymous / Private"
- **Off-net call forwarding**: When an inbound call is placed to a station with forwarding enabled and the forwarding number is an off-net PSTN number, RTP Keepalives on LAN 2 must be enabled or the call will complete, but receive dead-air. LAN2 is the interface typically used to connect to the SIP Service Provider.**Twinning: RTP Keepaliv**es on LAN2 must be enabled for proper functioning. LAN2 is the interface typically used to connect to the SIP Service Provider.
- **Call display update**: Call display was not properly updated on PSTN phone to reflect **the tru**e connected party on calls that are transferred to the PSTN from the enterprise. After the call transfer was completed, the PSTN phone showed the party that initiated the transfer instead of the actual connected party.
- **IP Softphone** – must have **Call Waiting On** and **Offhook Station** checked on the **User →Telephony →Call Settings** page to allow for proper transfer and forwarding capabilities.
- **SIP REFER** – Windstream does not support **SIP REFER** at this time, but will in a later release, it is therefore recommended that **REFER** not be enabled. **REFER** is enabled by checking "**REFER Support**" under **Line→17** in the reference configuration.

## 2.2. Support

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

For technical support on Windstream SIP Trunking, contact Windstream using the Customer Service links at www.windstream.com.

## 3. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows an enterprise site connected to Windstream SIP Trunking.

Located at the enterprise site is an Avaya IP Office 500 V2 with ATM Combination Card (6210) and Digital Station Base Card (TCM8). The LAN1 port of Avaya IP Office is connected to the enterprise LAN while LAN2, the WAN port, is connected to the public network. Endpoints include an Avaya 1140E Series (SIP firmware) registered to IP Office, Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 1400 Series Digital Telephone, an Avaya T7100 Digital Telephone (Legacy Nortel), an Avaya 6210 Analog Telephone, an Avaya IP Office Softphone, and Avaya IP Office Phone Manager with embedded voicemail.

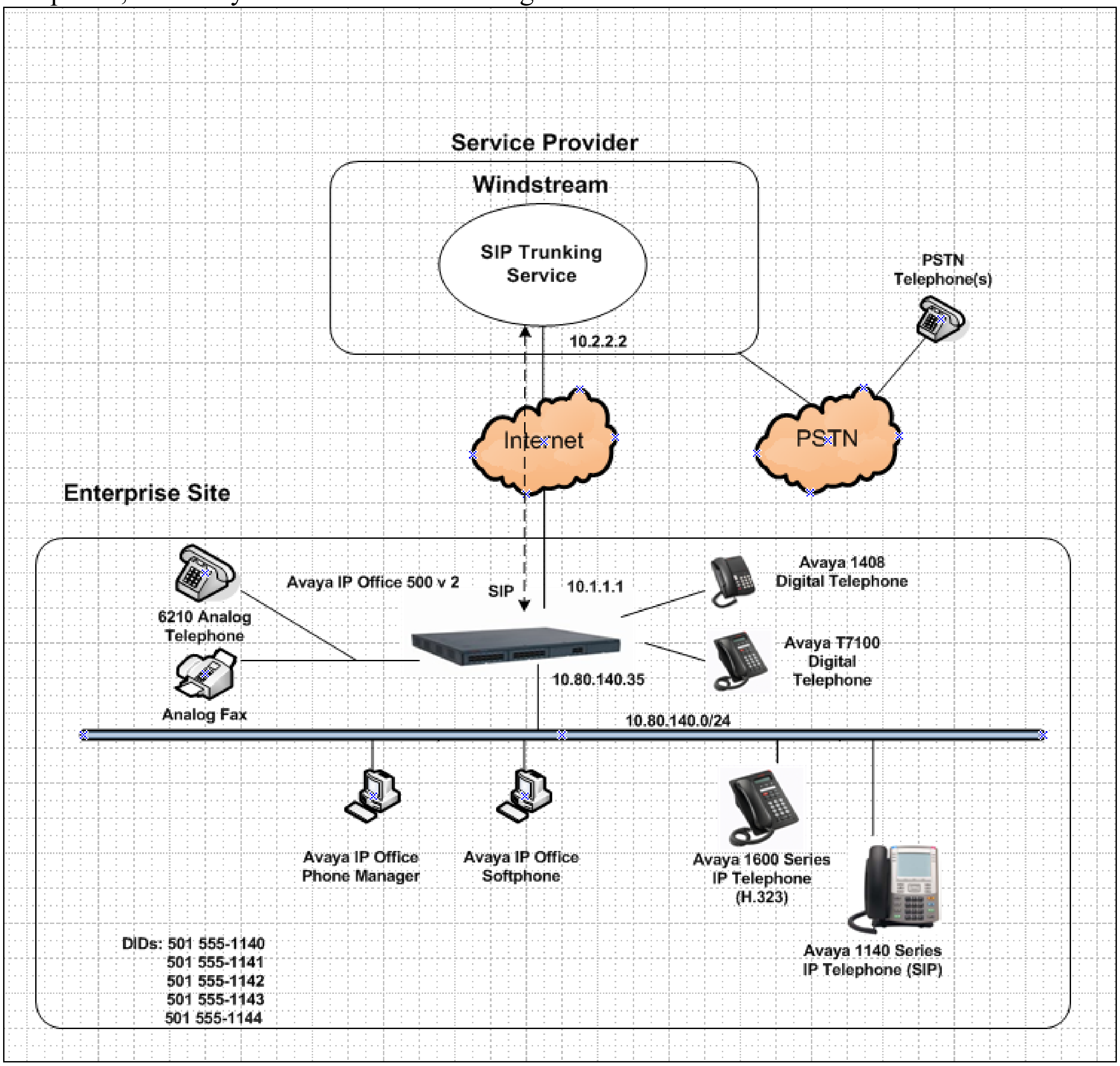



**Figure 1: Avaya IP Telephony Network Connecting to Windstream SIP Trunking Service**

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 cannot 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 Windstream. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Windstream. 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 field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, Windstream SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

Windstream uses the phone number in the From header of a SIP INVITE message to authenticate the calling party. Thus, a call will be rejected by the network unless the From header contains a number known to Windstream. This is especially important for calls inbound from the PSTN which are redirected back to the PSTN by call forwarding or twinning. For call forwarding, Avaya IP Office always sends the number of the forwarding phone in the From header. This is a number known to Windstream. As a result, the call display on the destination phone shows the forwarding party not the original caller. For twinning, this behavior can be slightly altered through configuration. See **Section 5.4** and **5.5.3** for details.

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

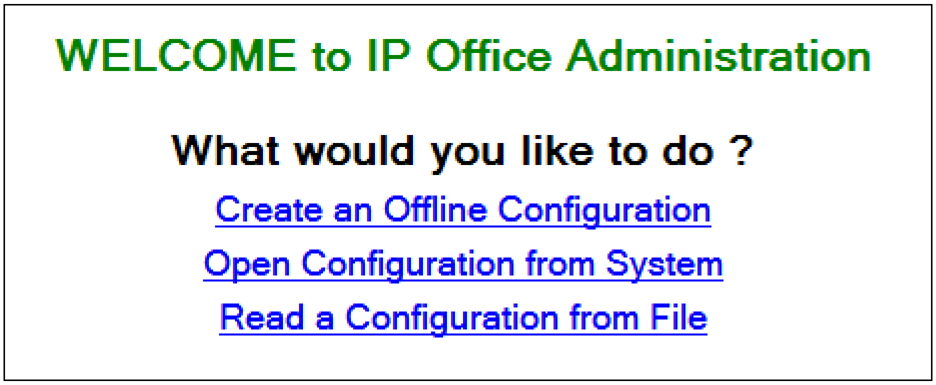
The following equipment and software were used for the sample configuration provided:

| **Avaya IP Telephony Solution Components** | |
|---|---|
| Component | Release |
| Avaya IP Office 500 v2<br>ATM Combination Card (6210)<br>Digital Station Base Card (TCM8) | 7.0.5 |
| Avaya 1140E Series IP Telephone (SIP) | 04.01.13.00 |
| Avaya 1616L Series IP Telephone (H323) | 1.3000 |
| Avaya 1408 Digital Telephone | n/a |
| Avaya T7100 Digital Telephone | n/a |
| Avaya 6210 Analog Telephone | n/a |
| Avaya IP Office Softphone (SIP) | 3.1.2.17 (59616) |
| **Windstream *Voice and Data Bundle – SIP* Components** | |
| **Component** | **Release** |
| Metaswitch - CFS | 7.1.01-B48 P90.41 |
| Metaswitch - UMG | 7.1.01-SU64 P86.00 |
| Metaswitch - EMS | 7.3.00-SU16 P86.00 |
| Adtran – TA908E | A02.06 |

**Table 1: Equipment and Software Tested**

# 5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to Windstream SIP Trunking. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a 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. A screen that includes the following in the center may be displayed:

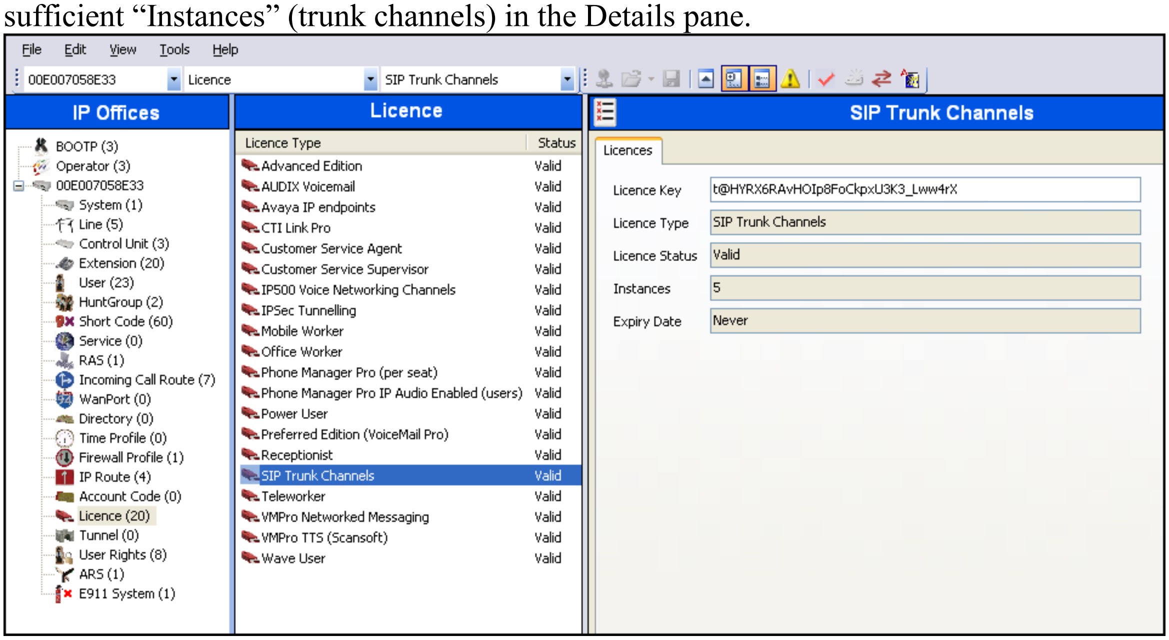



Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side.

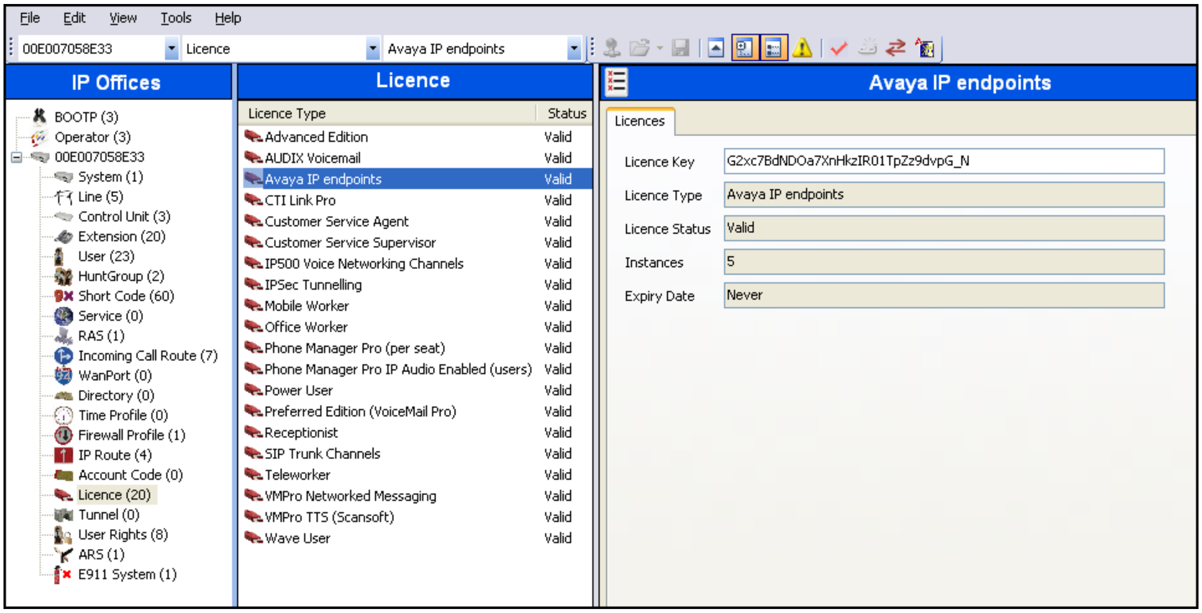
## 5.1. Licensing and Physical Hardware

The configuration and features described in these Application Notes require the IP Office system 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 and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient "Instances" (trunk channels) in the Details pane.




If Avaya IP Telephones will be used as is the case in these Application Notes, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient "Instances" in the Details pane.



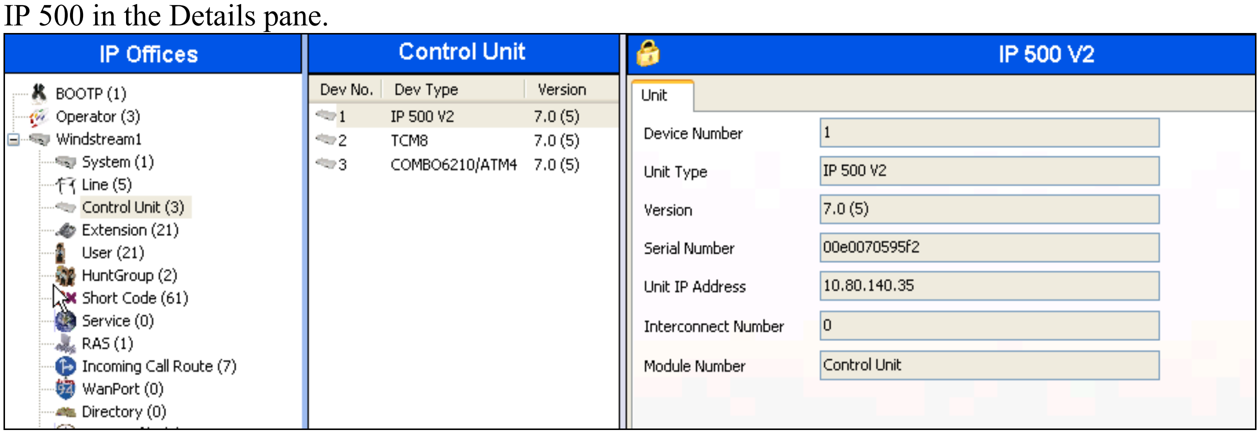

## 5.2. Physical, System and Network Settings

This section describes attributes of the sample configuration, but is not meant to be prescriptive. Consult reference [IPO-INSTALL] for more information on the topics in this section.

### 5.2.1. Physical Configuration and Settings

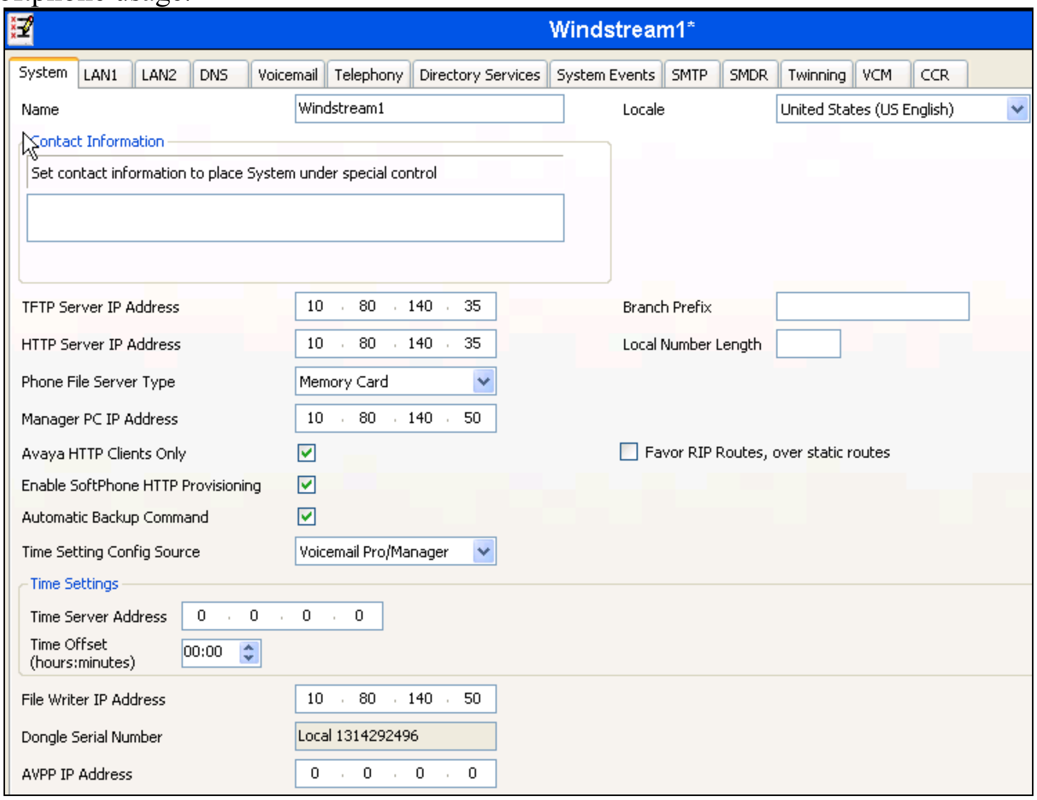
In the sample configuration, looking at the IP Office 500 v2 from left to right, the first module is the TCM8 card, a digital station module, the next slot from left to right contains the COMBO 6210/ ATM4 card, an analog and digital combo card, and the last slot was blank (i.e., no module is physically inserted). The TCM8 module allows connection of legacy Nortel digital stations, a T7100 was used in this configuration. The COMBO6210/ATM module allows connection of analog and Avaya digital endpoints. In the testing of the sample configuration, an analog telephone or a fax machine is connected to port 8 of the module and an Avaya 1408 digital station is connected to port 1.

The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. In the screen below, **IP 500 V2** is selected in the Group pane, revealing additional information about the IP 500 in the Details pane.



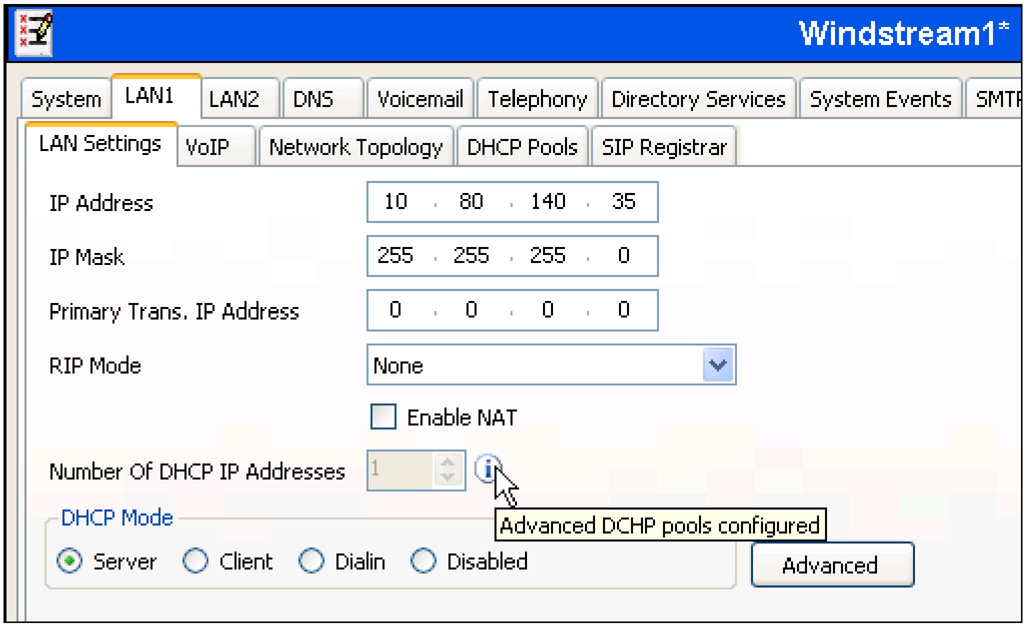
### 5.2.2. System Settings

Select **System** from the Navigation pane to access the main system menu. The **Name** field can be used for a descriptive name of the system, **Windstream1** is used here. The TFTP and HTTP server fields indicate the Avaya IP Office LAN1 IP Address for endpoints to discover. The **Avaya HTTP Clients Only** and **Enable SoftPhone HTTP Provisioning** boxes are checked to facilitate Avaya IP Office Softphone usage.

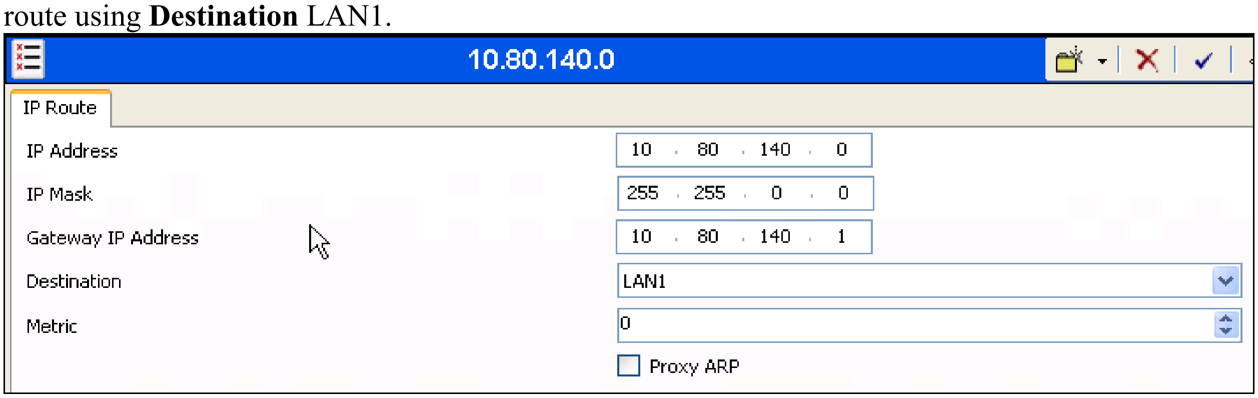



### 5.2.3. LAN1 Settings

Under **System→LAN1→LAN Settings**, the **IP Address** of the LAN Interface is set, along with the **IP Mask** of the subnet and the **DHCP Mode**. Here the LAN1 IP Address is **10.80.140.35** with a mask of **255.255.255.0** and the **DHCP Mode** is set to **Server**. The indicates that Advanced DHCP settings are being used and by rolling the mouse over the button you can see the notice. The **DHCP Pools** tab will display the DHCP config (not shown).

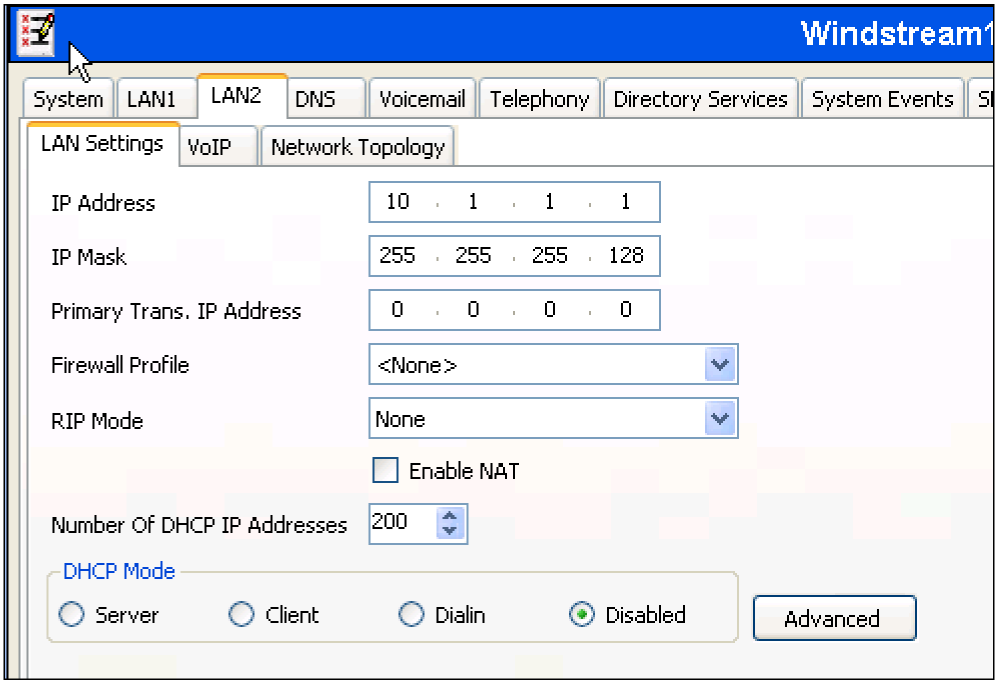



In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.80.140.1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New.** To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant default route using **Destination** LAN1.

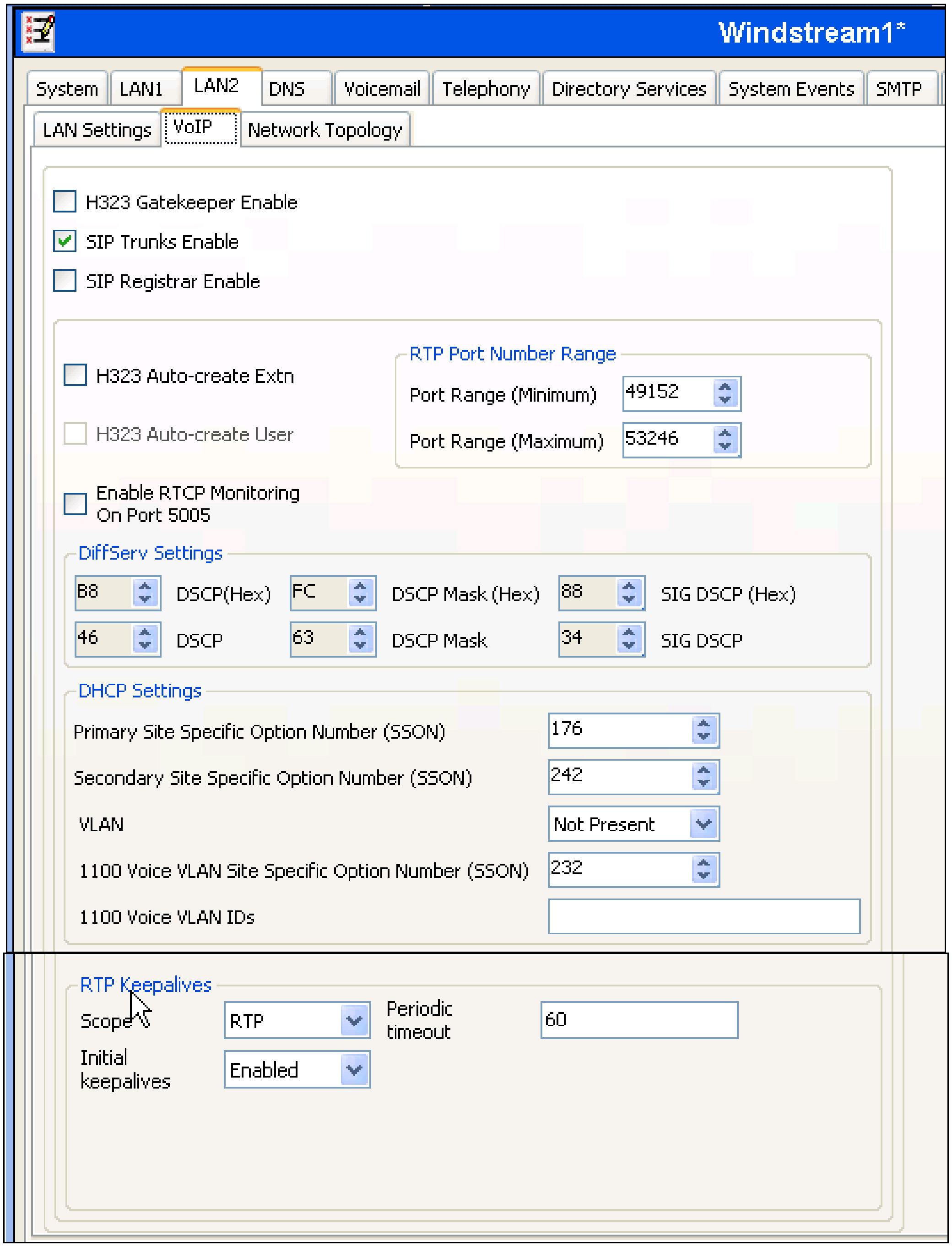



### 5.2.4. LAN2 Settings

In the sample configuration, 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 **System →Windstream1** 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. **RTP Keepalives** must be enabled or Forwarding Off-net and Twinning does not function properly. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.

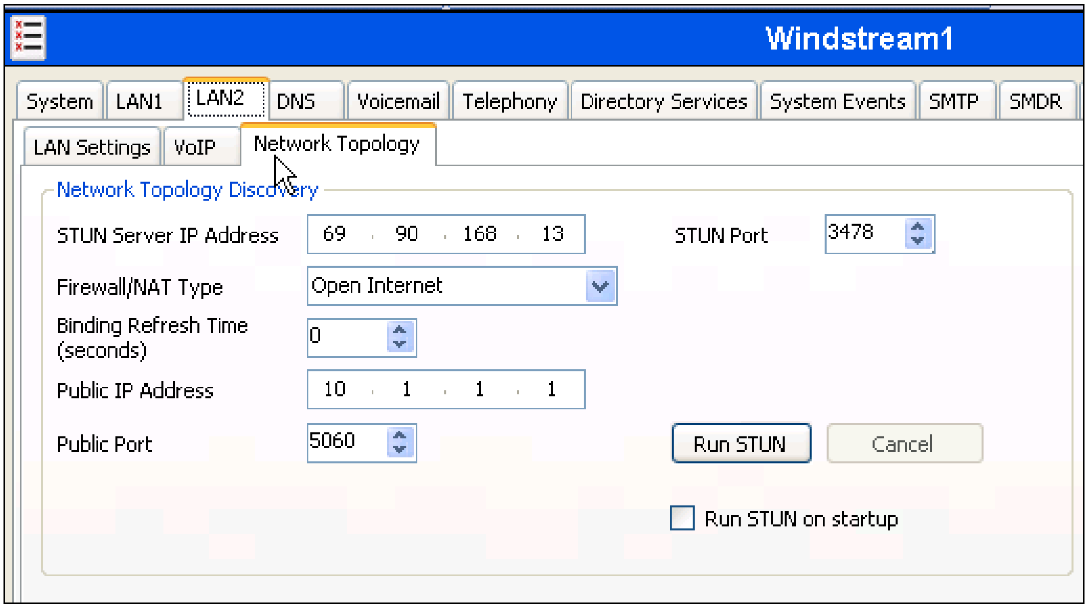



On the **Network Topology** tab in the Details Pane, configure the following parameters:

- 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 ***0***. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS

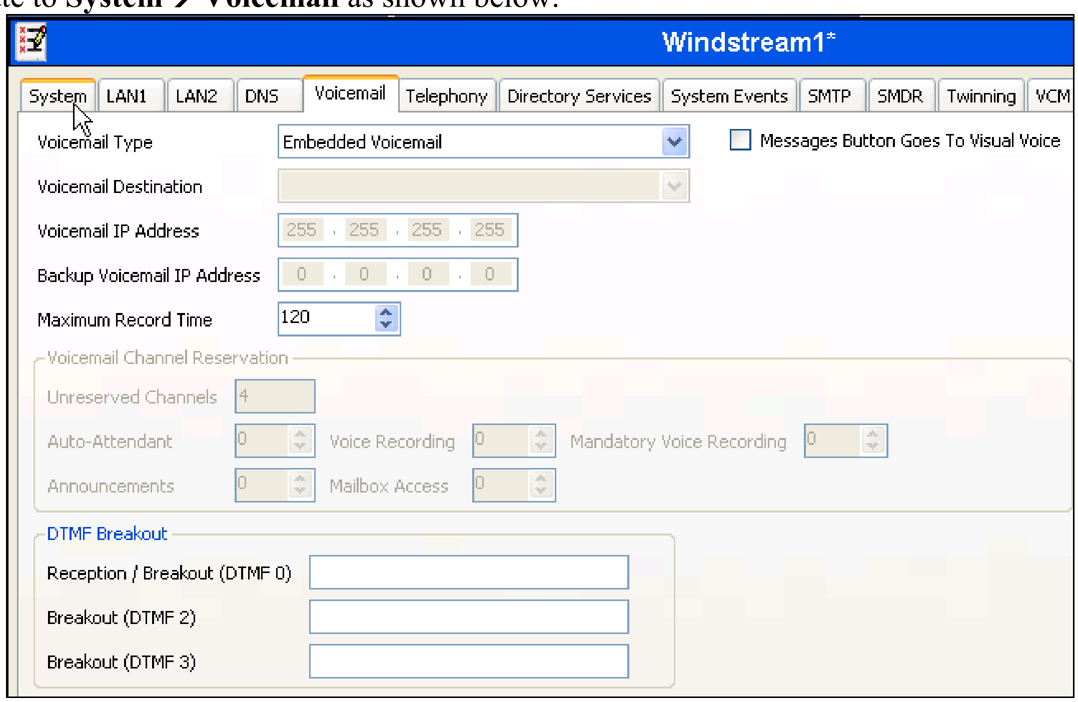
messages to the service provider. Windstream responds to OPTIONS messages with a **501 Not Implemented**, so this was set to ***0***, which means that **OPTIONS** messages are sent only every 5 minutes.

- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- All other parameters should be set according to customer requirements.



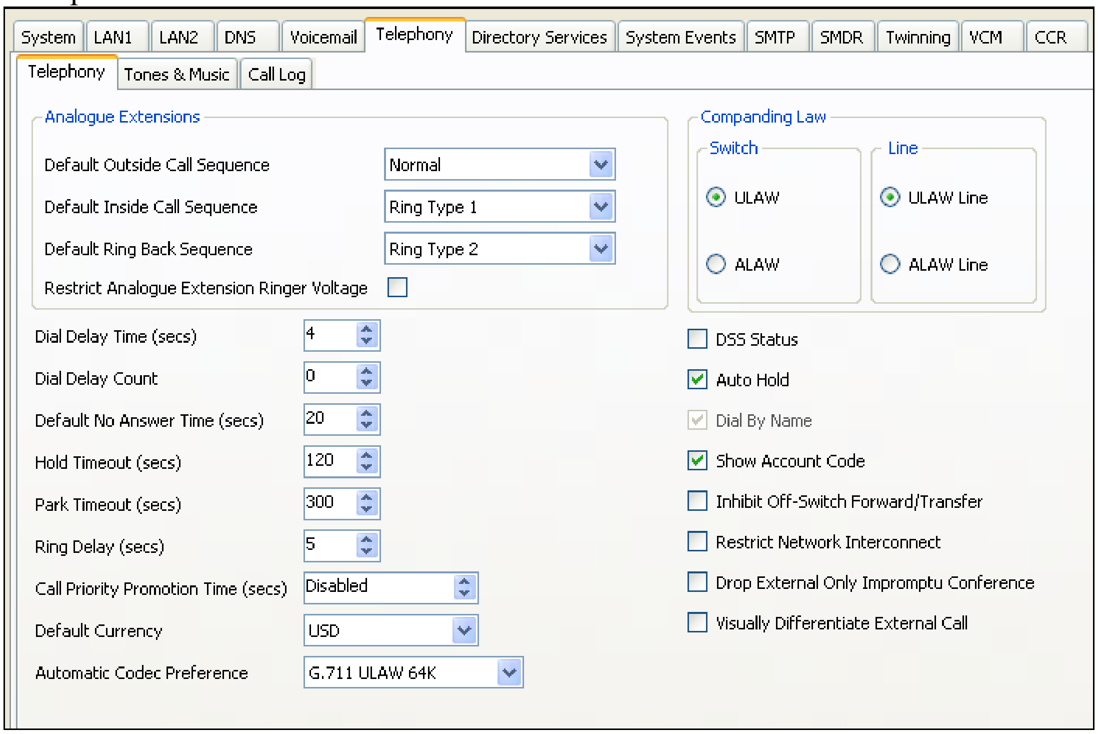

## 5.3. Voicemail

In the sample configuration, **Embedded Voicemail** was used. To change voicemail settings, navigate to **System→ Voicemail** as shown below.



## 5.4. System Telephony Settings

Navigate to the **Telephony →Telephony** Tab on the Details Pane. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. Choose the **Companding Law** typical for the enterprise location. For North America, ***ULAW*** 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.

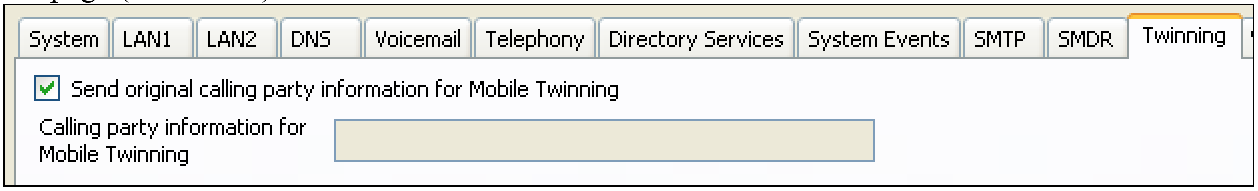



## 5.5. Twinning Calling Party Settings

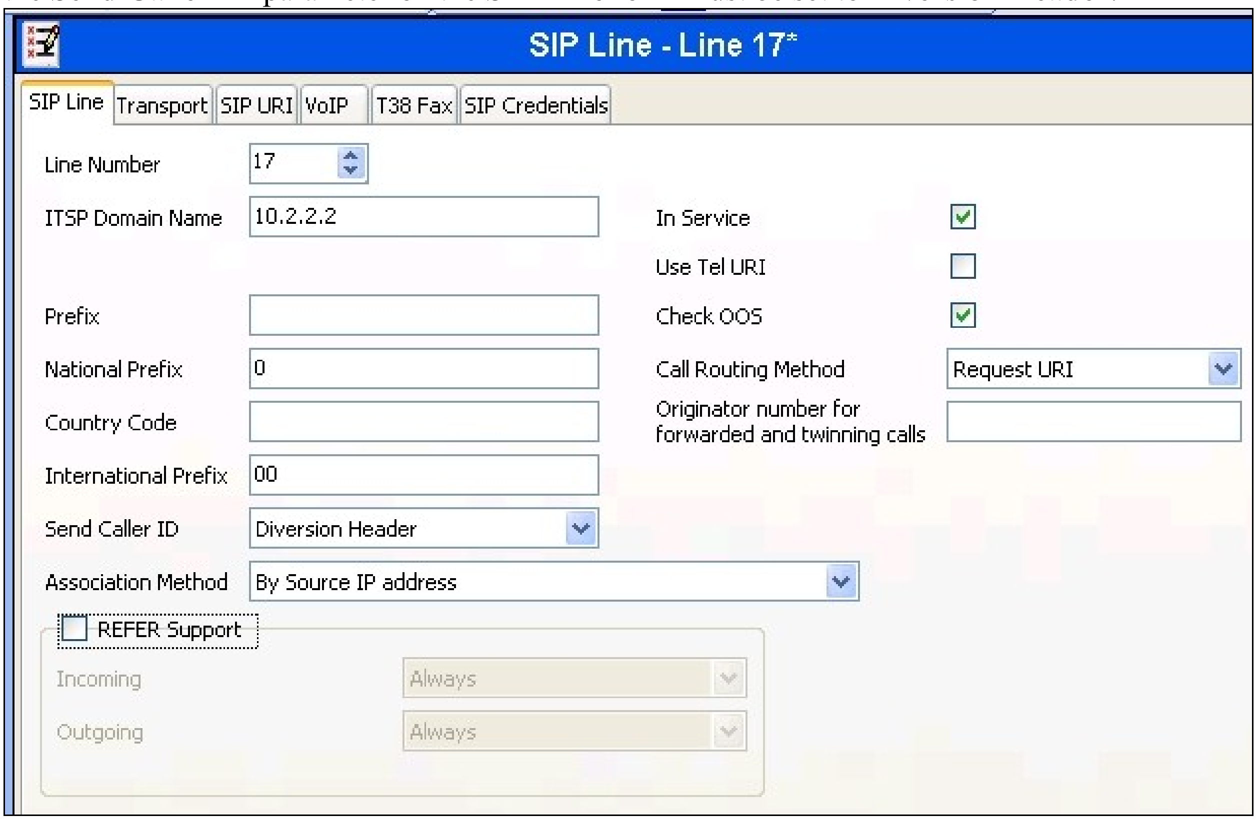
When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affect twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.6**). If this box (representing the first parameter) is checked, the setting of the second parameter is ignored and Avaya IP Office will send the following in the SIP From Header:

- On calls from an internal extension to a twinned phone, Avaya IP Office will send the calling party number of the originating extension.
- On calls from the PSTN to a twinned phone, Avaya IP Office will send the calling party number of the host phone associated with the twinned destination (instead of the number of the originating caller).

If this box is unchecked, the value sent in the SIP From header is determined by the setting of the second parameter. For the compliance test, the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab was checked which overrides any setting of the **Send Caller ID** parameter on the **SIP Line** form.  Click the **OK** Button at the bottom of the page (not shown).




To have the caller-id displayed on the twinned phone be the originating caller instead of the host phone associated with the twinned destination on a PSTN call, the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab must be unchecked and the **Send Caller ID** parameter on the **SIP Line** form must be set to **Diversion Header**.



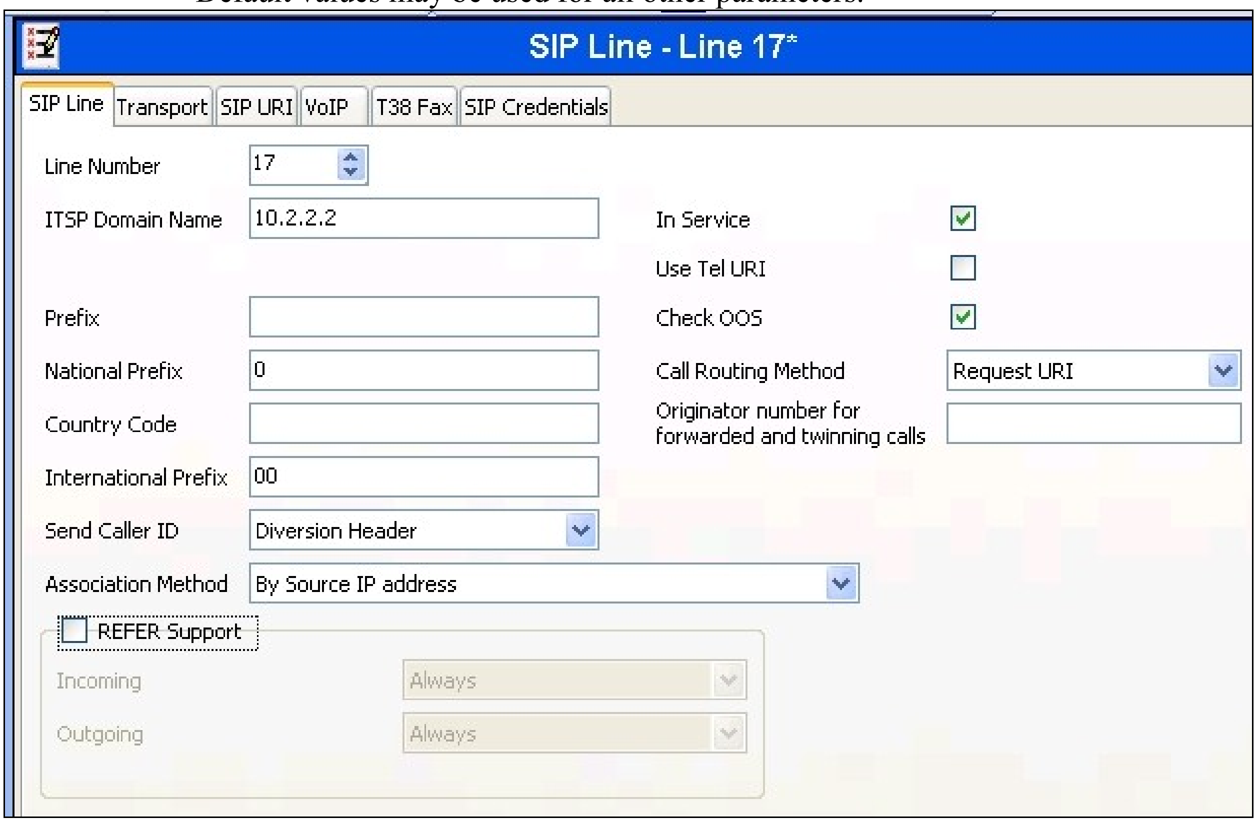

## 5.6. Administer SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 7. Since IP Office Release 7 introduced new SIP Line parameters and re-oriented existing parameters, this section has the most substantive changes in these Application Notes.

### 5.6.1. SIP Line - SIP Line Tab

A SIP line is needed to establish the SIP connection between Avaya IP Office and Windstream SIP Trunking. 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 as shown below.

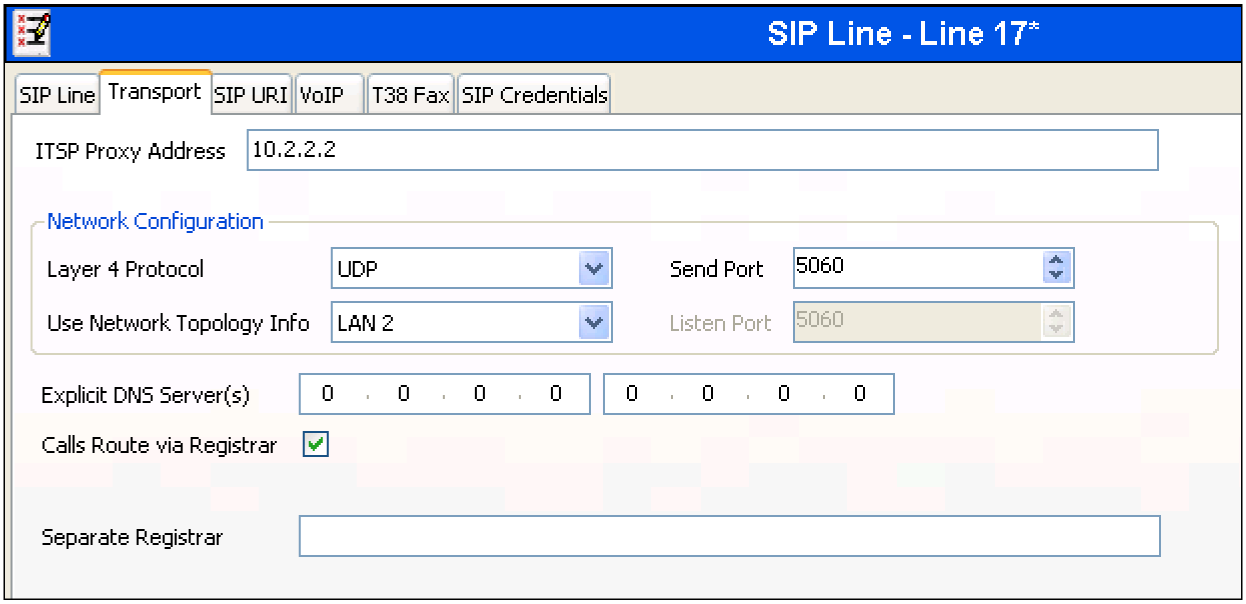
- Set **ITSP IP Address** to the IP address of the Windstream SIP proxy.
- Set **Send Caller ID** to ***None***. For the compliance test, this parameter was ignored since the **Send original calling party information for Mobile Twinning** box is checked in **Section 5.5**
- Check the **In Service** box.
- Uncheck the **Check OOS** box since OPTIONS messages are not supported. This MUST be unchecked or IP Office will never consider the trunk in service since the OPTIONS messages are not being responded to properly.
- UnCheck **REFER Support** since Windstream does not support **REFER** at this time.
- Default values may be used for all other parameters.




## 5.6.2. SIP Line - Transport Tab

Select the **Transport** tab. This tab was introduced in Release 6.1. Some information configured in this tab had been under the **SIP Line** tab in Release 6.0.

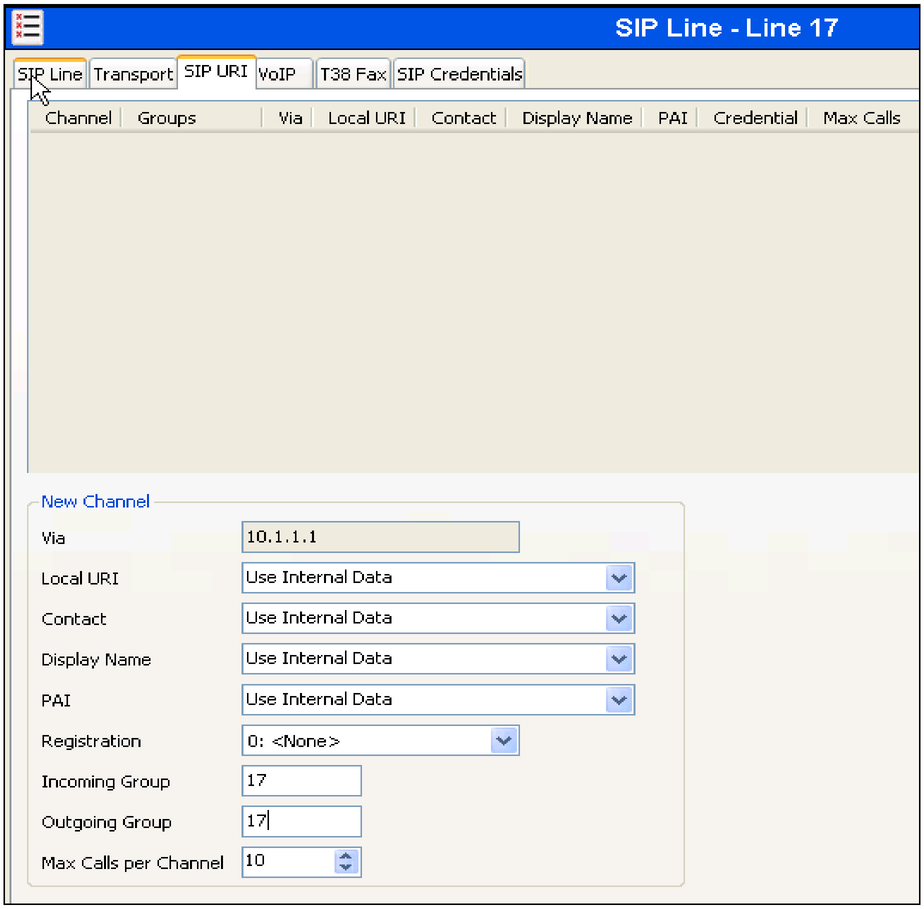
The **ITSP Proxy Address** is set to the IP Address provided by Windstream. As shown in **Figure 1**, this IP Address is 10.2.2.2. In the **Network Configuration** area, UDP is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Windstream. The **Use Network Topology Info** parameter is set to "LAN 2". This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab.




### 5.6.3. SIP Line - SIP URI Tab

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. 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 Avaya IP Office user. The entry was created with the parameters shown below.

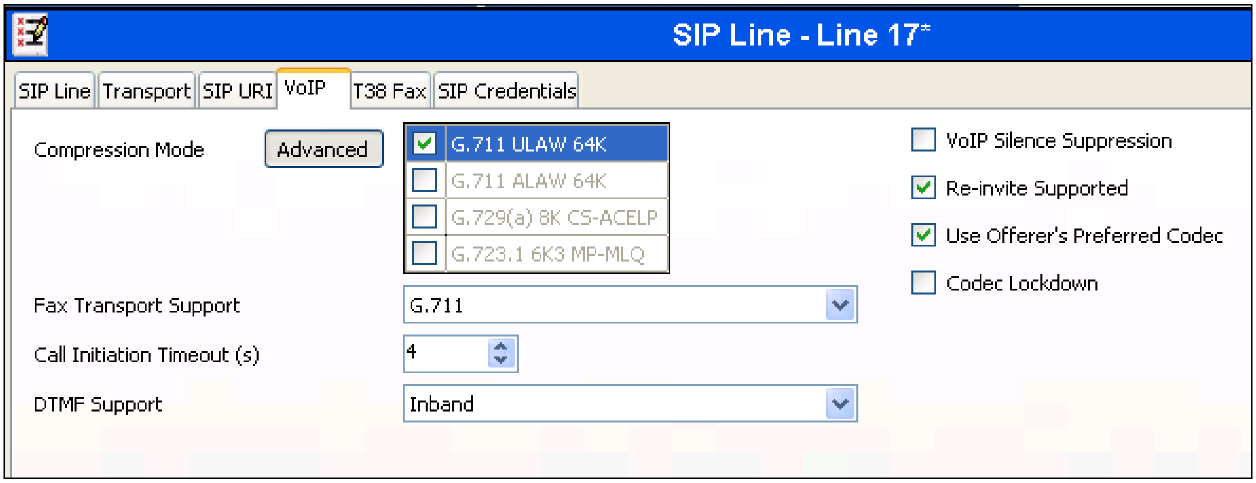
- Set **Local URI**, **Contact, Display Name,** and **PAI** to ***Internal Data***. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.8.1**.
- Associate this line with an incoming line group by entering a line group 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.




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

- Set the **Compression Mode** to ***G.711 ULAW 64K***. Windstream only supports G.711ULAW at this time.
- Set **Fax Transport Support** box to **G.711**. T. 38 is not supported at this time.
- Set the **DTMF Support** field to **Inband**. This is the only option Windstream supports.
- Check the **Re-invite Supported** box.
- Check the **Use Offerer's Preferred Codec.** Normally for SIP calls, the initiator of the calls sends a SIP offer which includes the codecs that they support in order of preference. The SIP response includes the codec they want to use for the call. This option can be used to override that behaviour and use the codec preference offered by the caller.
- Default values may be used for all other parameters.

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

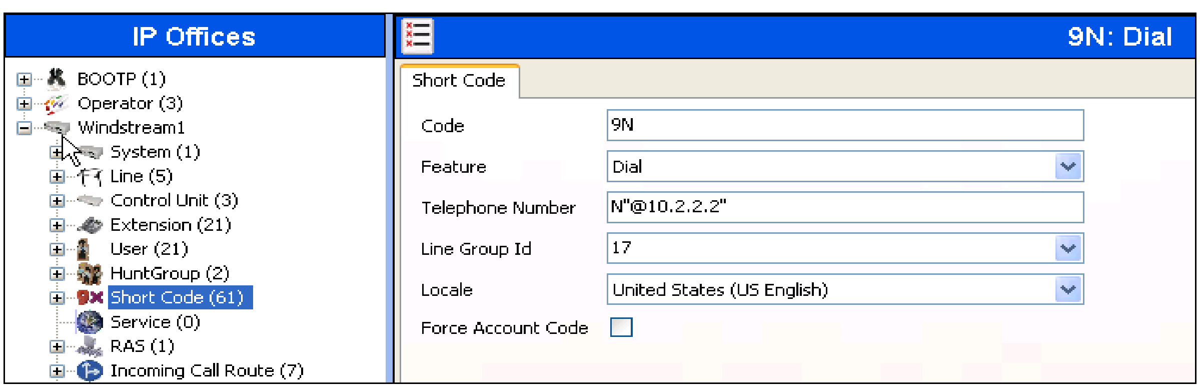



## 5.7. Short Code

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**. 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 semicolon. 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"@10.2.2.2"***. 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 10.2.2.2 is the IP address of the Windstream SIP proxy.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6.1**. This short code will use this line group when placing the outbound call.

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

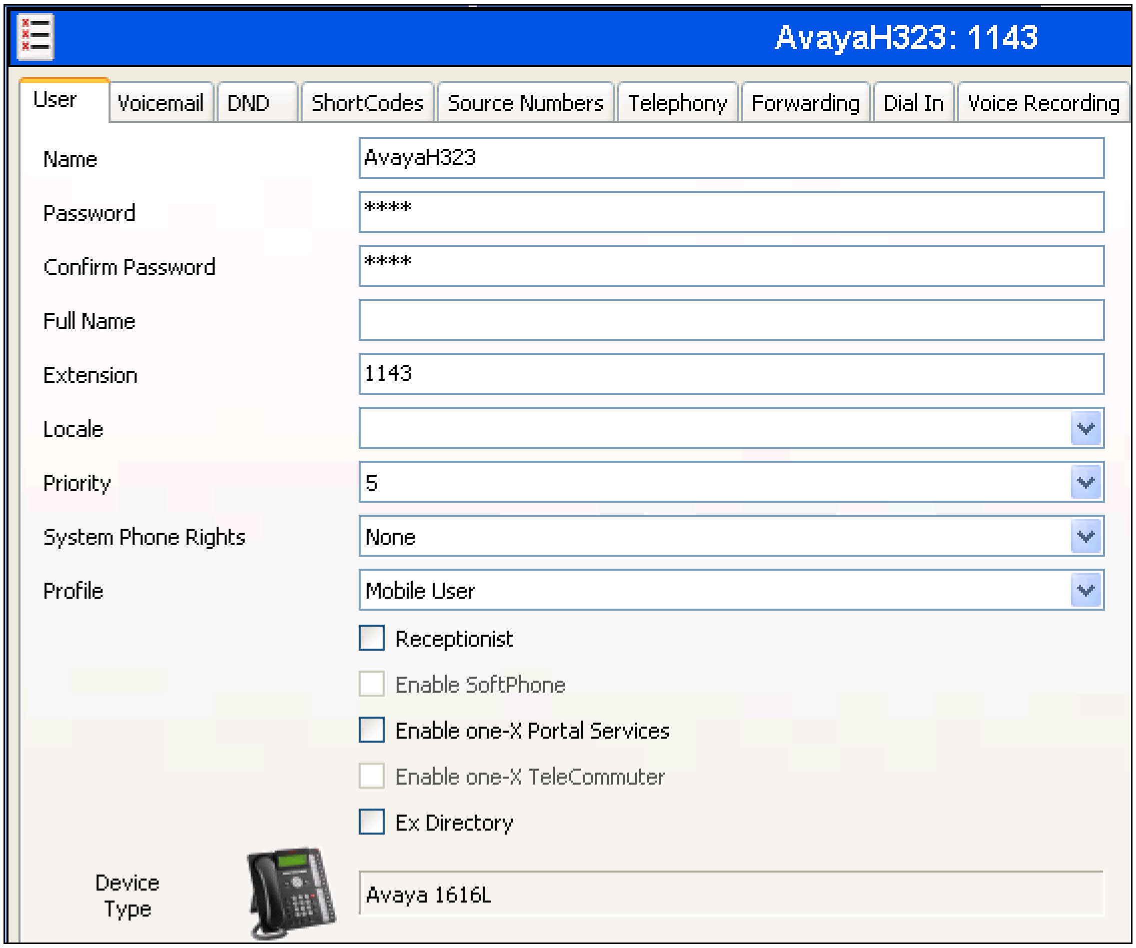



## 5.8. User

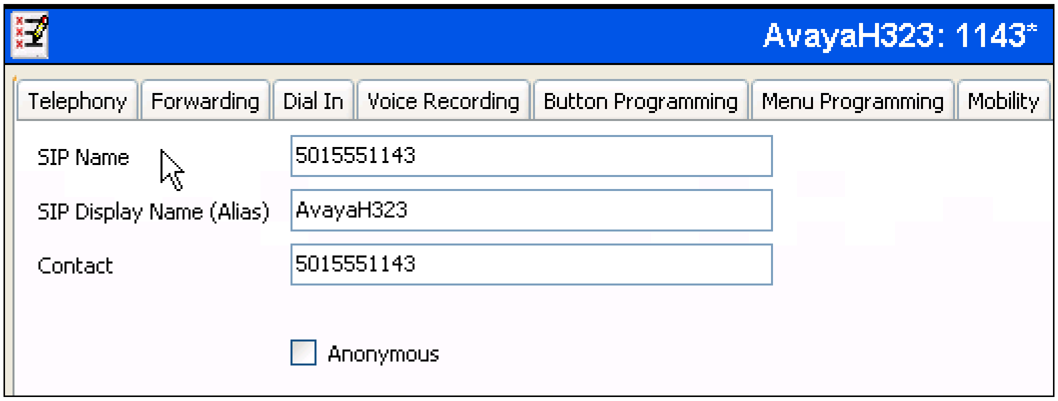
In this section, examples of IP Office Users and Extensions will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the Navigation pane, and select **New.** To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane.

### 5.8.1. User 1143

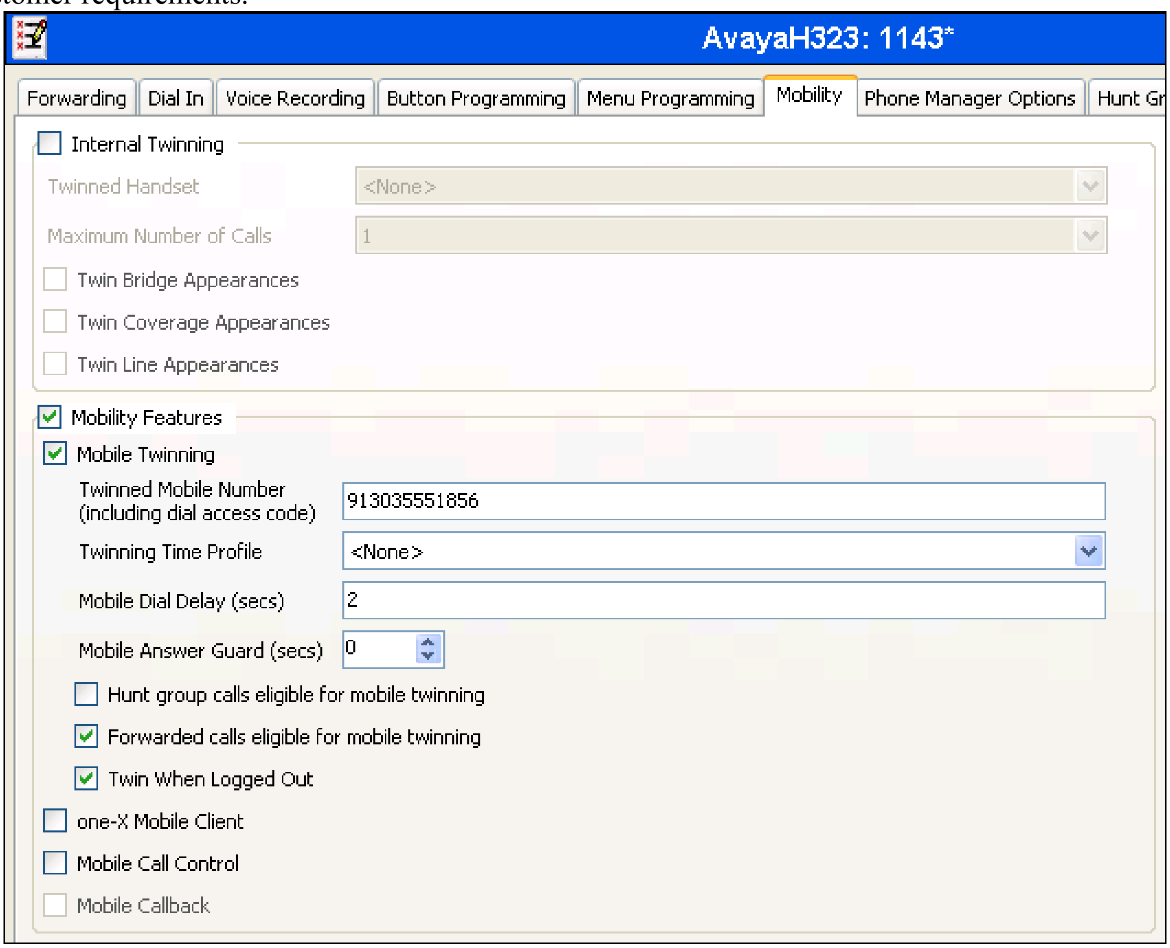
The following screen shows the **User** tab for User 1143. This user corresponds to the H.323 telephone 1616L.




The following screen shows the **SIP** tab for User 1143. The **SIP Name** and **Contact** parameters are configured with a Windstream SIP Trunk DID number for the user, 5015551143. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls from Windstream SIP Trunk Service, without having to enter this number as an explicit SIP URI for the SIP Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. **Anonymous** <u>cannot</u> be checked to withhold user information as Windstream does not allow call-id restrictions on outgoing calls.




User 1143 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 1143. The **Mobility Features** and **Mobile Twinning** boxes are checked**.** The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 913035551856, with **9** being a short code, see **Section 5.7**. Other options can be set according to customer requirements.

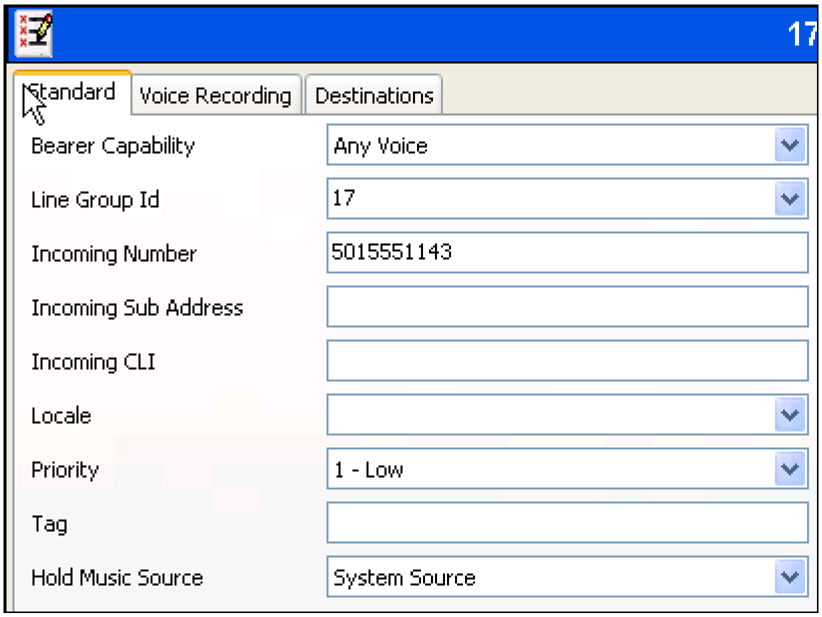



## 5.9. Incoming Call Route

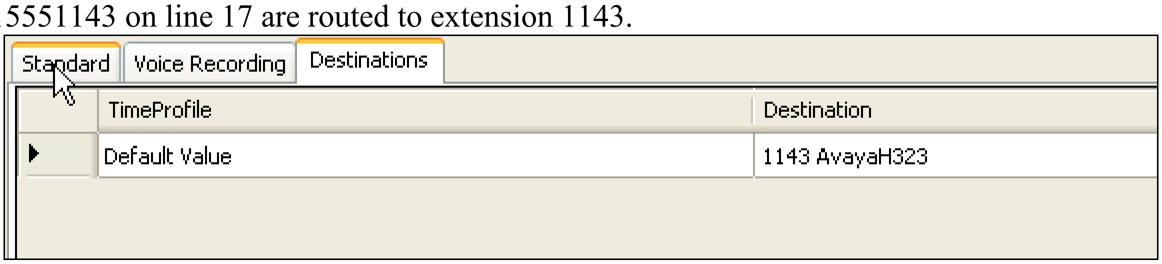
An incoming call route maps an inbound DID number on a specific line to an internal extension.

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**. 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 5.6.1**
- 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.

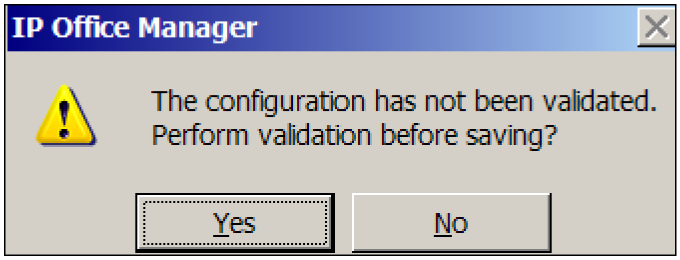



On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 5015551143 on line 17 are routed to extension 1143.

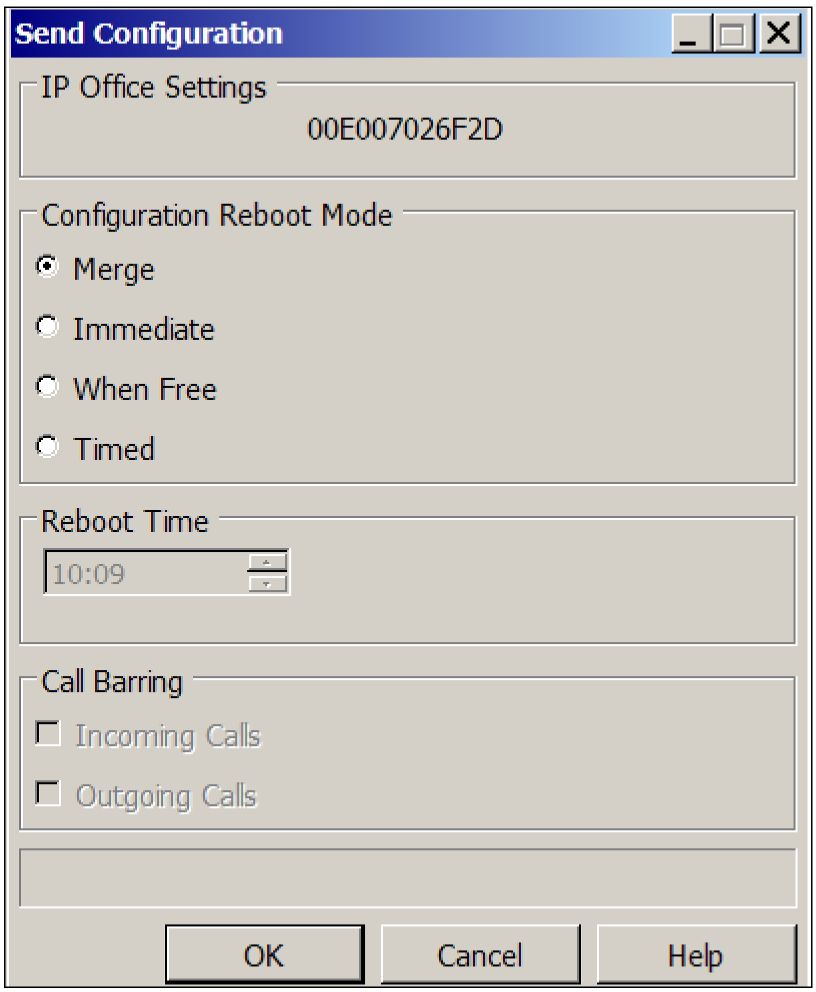



## 5.10. Save Configuration

When desired, send the configuration changes made in IP Office Manager to the IP Office server, to cause the changes to take effect. Click the "disk" icon that is the third icon from the left (i.e., common "save" icon with mouse-over help "Save Configuration File"). Click **Yes** to validate the configuration, if prompted.




Once the configuration is validated, a screen similar to 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** if desired.




# 6. Windstream SIP Trunking Configuration

Windstream is responsible for the configuration of Windstream SIP Trunking. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. This is the LAN2 address configured in **Section 5.2.4**. Windstream will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Windstream including:

- IP address of the Windstream SIP proxy

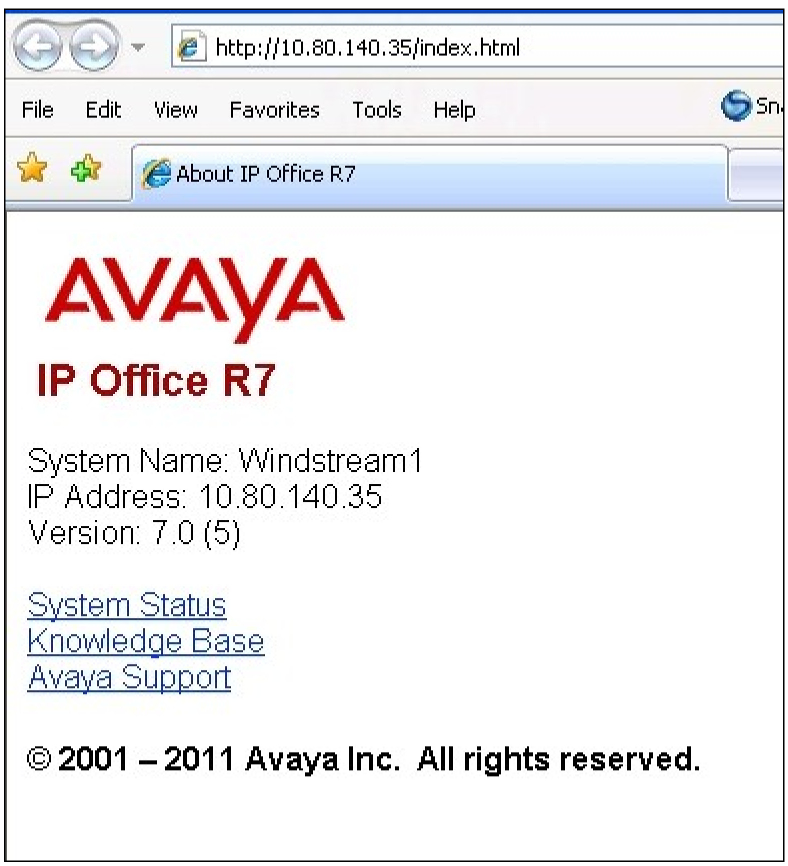
- Supported codecs
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

# 7. Verification Steps

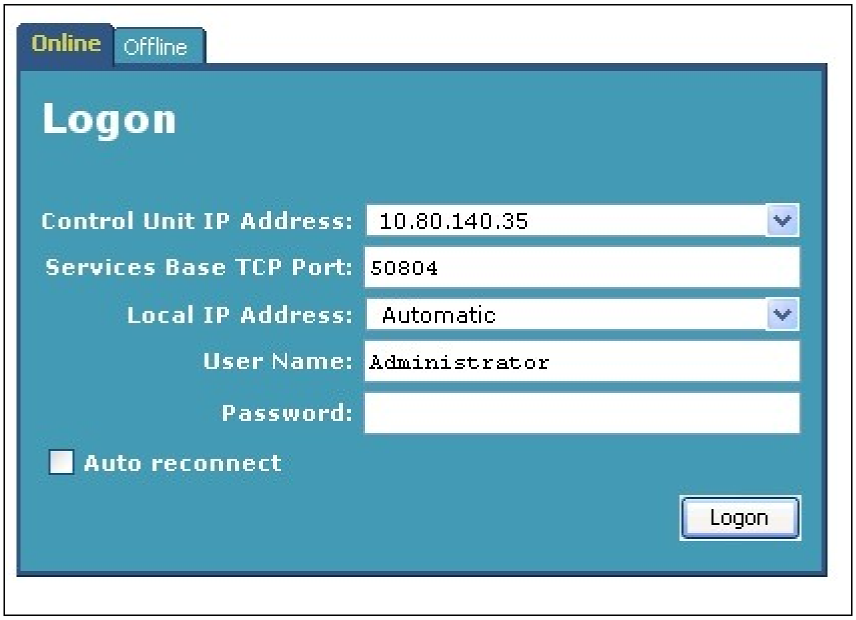
This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

## 7.1. System Status

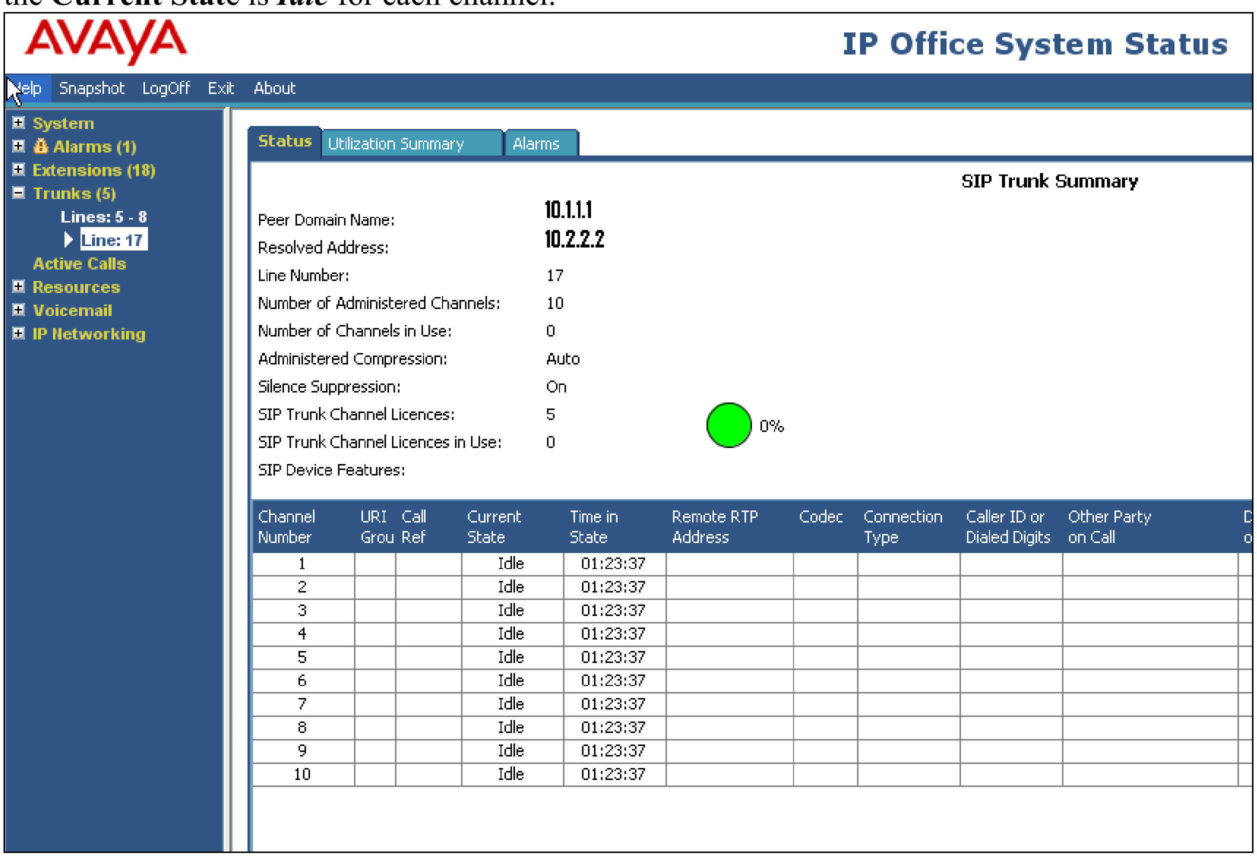
The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start →Programs →IP Office →System Status**. Or by opening an Internet browser and type the URL: http://*ipaddress* where *ipaddress* is the IP address of the Avaya IP Office LAN1 interface. Click on **System Status** to launch the application.



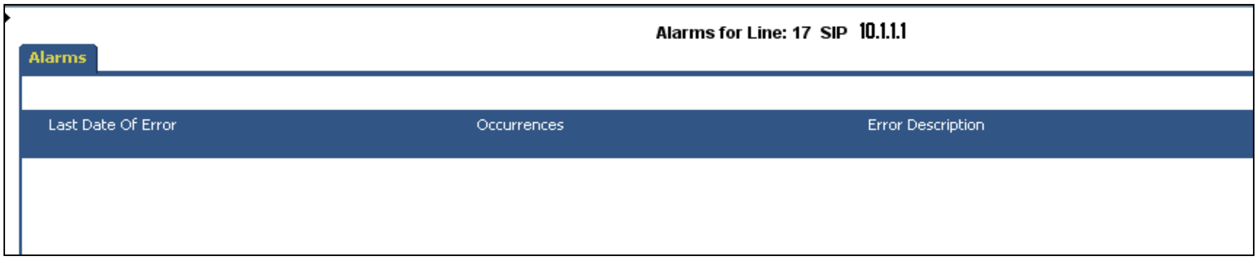

The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.




Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current Stat**e is ***Idle*** for each channel.



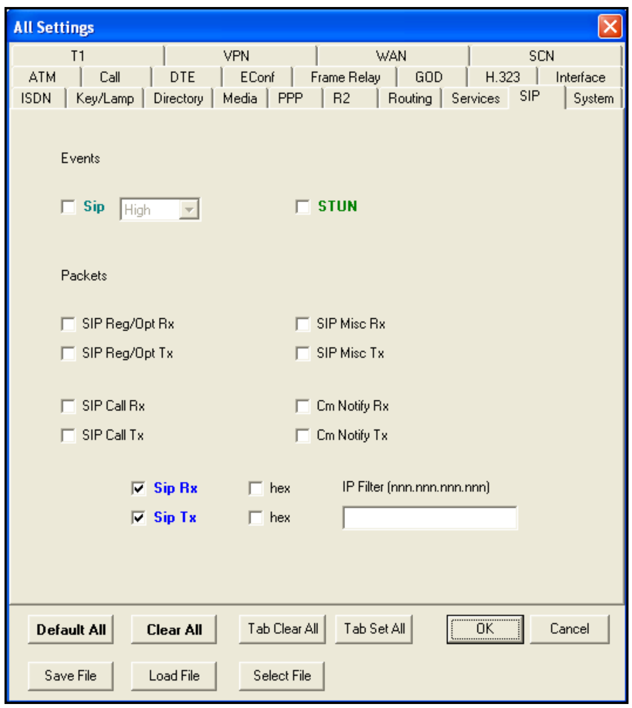

Select the **Alarms** tab and verify that no alarms are active on the SIP line.



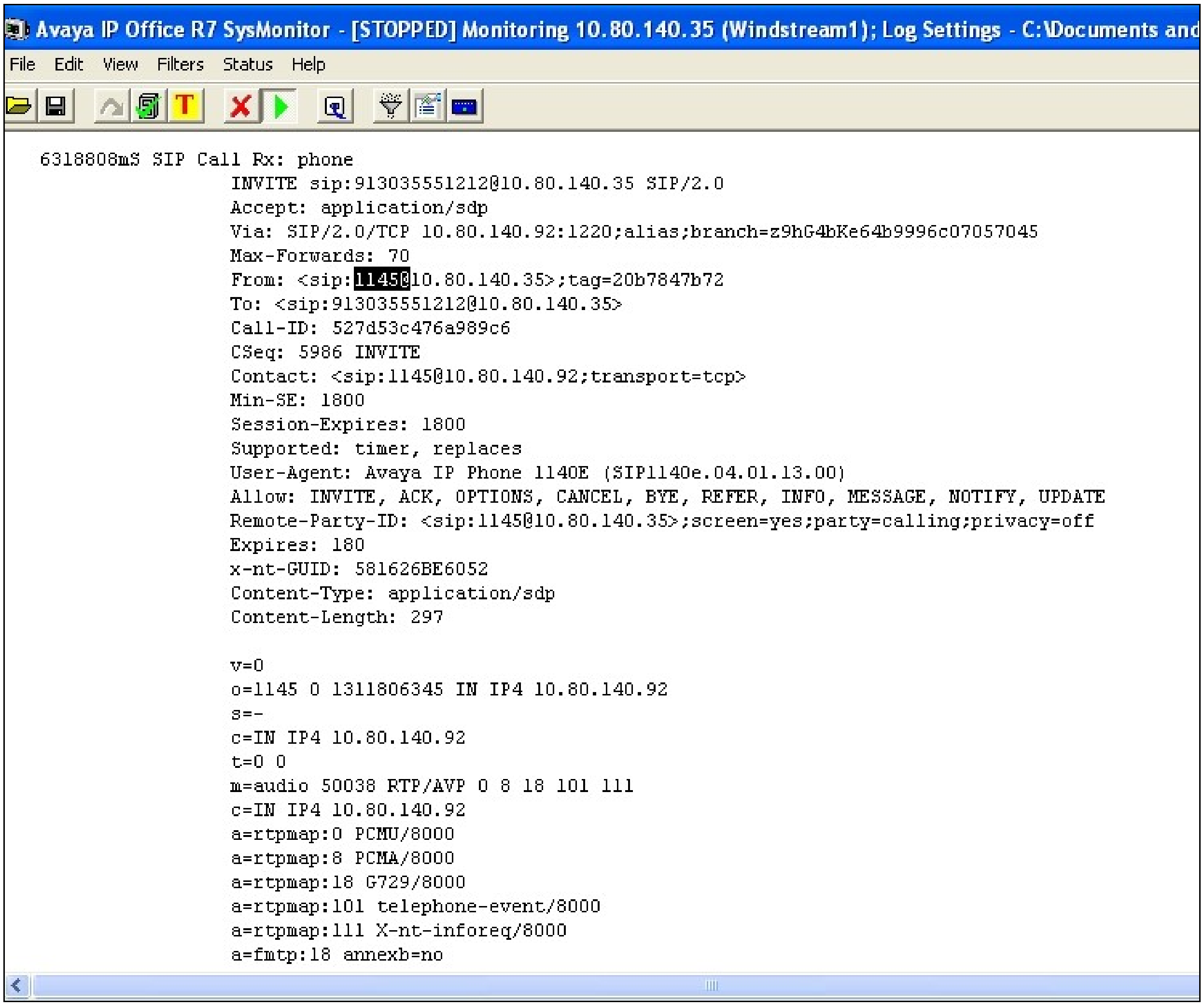

## 7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start** →**Programs** →**IP Office** →**Monitor**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters** →**Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.




As an example, the following shows a portion of the monitoring window for an outbound call from extension 1145, whose DID is 501-555-1145, calling out to the PSTN via the Windstream IP Trunking Service. The telephone user dialed 9-1-303-555-1212.




- 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

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

## 9. References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1][IPO-INSTALL] IP Office 7.0 Installation Manual, Document Number 15-601042, Issue 23e (March 16, 2011).

[2]*IP Office 7.0, IP Office Standard Version Installation*, 15-601042 Issue 23k - (22 May 2011).
[3]*IP Office Release 7.0*, *1100/1200 Series Phone Installation,* Issue 01c - (22 March 2011).
[4]*IP Office, H323 IP Telephone Installation*, 15-601046 Issue 17a - (07 March 2011).
[5]*IP Office Release 7.0, Manager 9.0,* Release 6.0, 15-601011 Issue 26h - (22 May 2011).
[6]*IP Office, IP Office System Monitor*, 15-601019 Issue 02b - (28 November 2008).
[7]*IP Office, SIP Extension Support*, Issue 1c - (20 February 2010).
[8]*IP Office Release 7.0, IP Office Softphone Installation,* Issue 2b - (21 February 2011).
[9]RFC 3261 *SIP: Session Initiation Protocol,* http://www.ietf.org/
[10]RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, http://www.ietf.org/
[11]RFC 4244, An Extension to the Session Initiation Protocol (SIP) for Request History Information, http://www.ietf.org/

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for Windstream SIP Trunking is available from Windstream.

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.