Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office 7.0 Integration with Skype Connect R2.0 – Issue 1.0

Abstract

These Application Notes describe the steps to configure an Avaya IP Office SIP trunk solution with Skype Connect R2.0. The Skype Connect R2.0 service referenced within these Application Notes is designed for business customers with an Avaya SIP trunk solution. The service provides bi-directional local and/or long distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

Testing was conducted at the Avaya Solution & Interoperability Test Lab utilizing a traditional Internet T1 ISP circuit for accessing the Skype Connect R2.0 service directly over the Internet.

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1. Introduction

These Application Notes describe the steps to configure the Avaya SIP trunk solution with Skype Connect using an Internet-based connection. Skype Connect enables a business to use their Skype Connect certified hardware to take advantage of Skype's global calling rates to landline and mobile phones. The service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks. Also, businesses may choose to purchase separately Skype's online numbers to receive calls. Access to a broadband Internet connection is required.

The Skype Connect service uses multiple session border controllers (also called service nodes) in the Skype network to deliver service redundancy. The Avaya SIP trunk architecture consists of Avaya IP Office (version 7.0) at the enterprise site with a SIP trunk configured to the Skype Connect service. While not the focus of this testing, Avaya IP Office Voicemail Pro was used to provide enterprise voicemail call coverage for Avaya telephones.

The configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed below.

- Response to SIP OPTIONS queries.
- Incoming calls to various IP Office supported phone types (i.e. H.323, digital, and analog telephones at the enterprise). All inbound calls were routed to the enterprise across the SIP trunk from Skype. Inbound calls were place from the PSTN and from Skype users.
- Outgoing calls from various IP Office supported phone types (i.e. H.323, digital, and analog telephones at the enterprise). All outbound calls were routed from the enterprise across the SIP trunk to Skype.
- Inbound and outbound long holding time call stability.
- Various call types including: local, long distance, international, outbound toll-free.
- Codec Negotiation.
- Caller ID presentation.
- Privacy requests (i.e., caller anonymity) and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as call park, call pickup, hold and resume, transfer and conference.
- Off-net call forwarding.

This document assumes that the installer has undergone Avaya approved training and has a working knowledge of IP Office installations.

1.1. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to the Skype Connect service through the public IP network.

An Avaya IP Office 500 system is located at the enterprise site. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network (i.e. internet). Endpoints include Avaya 5600 Series IP Telephones (with H.323 firmware), Avaya Avaya 2400 Series Digital Telephones, and an Avaya 6210 Analog Telephone. The site also has a Windows XP Service Pack 3 PC running the Avaya IP Office Manager application to configure Avaya IP Office. The same PC also runs Avaya Voicemail Pro for providing a voice messaging service to the Avaya IP Office users.

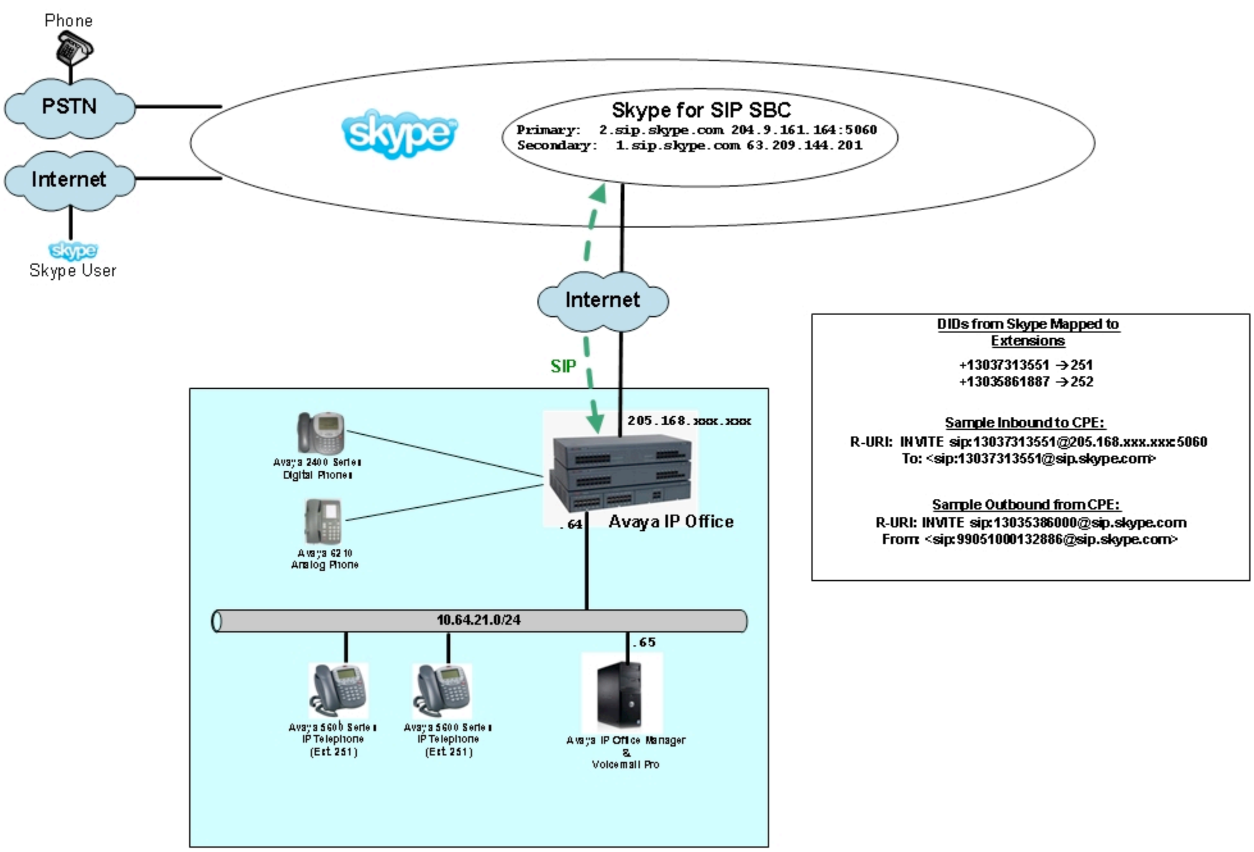


Figure 1: Avaya IP Office and Skype Connect

For the purposes of the interoperability testing, Avaya IP Office users dialed a short code of 9 + N digits to send digits across the SIP trunk to Skype. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Skype. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the ***To*** field of an outbound SIP INVITE message. For inbound calls, the Skype Connect server sent 11 digits in the Request URI and the ***To*** field of inbound SIP INVITE messages.

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

1.2. Known Limitations

The following limitations are noted for the reference configuration described in these Application Notes:

- Avaya IP Office 7.0 should currently only be deployed for SKYPE using IP authentication only. Otherwise, Avaya IP Office will "drop" registration after a period of time (varied). This issue is being tracked with Avaya tracking MR: MRDB00124075. A maintenance release will be available 1st quarter of 2012 that resolves a registration issue found in testing detailed in the following Avaya MR: MRDB00124075**.**
- **Outbound DTMF**: During testing, an IP Office phone called a PSTN number that prompted the caller to enter a meeting security code. After the far-end completed collecting the digits entered, it replayed the digits entered back to the caller. 10 digits were entered and detected by IP Office; however, only 9 digits were played back to the caller.
- **International Call with 011 prefix**: If an IP Office phone places an outbound international call without the 011 prefix being sent to Skype in the Request URI, the call succeeds. If the 011 prefix is sent to Skype, the call fails. Skype sends back a "484 Address Incomplete" message back to IP Office.
- **Outbound caller-id restriction**: The IP Office Caller ID blocking feature is not compatible with Skype using a short code with the Telephone Number field set to NW. To make anonymous calls using, enable the Anonymous checkbox on the UseràSIP tab with a short code Telephone Number of N. This issue was noted during the last round of testing with IP Office 6.1 and an Avaya tracking MR was created: MRDB00041191.
- Fax is not supported by Skype.
- Skype does not support Early Media in SIP.
- Inbound calls to Avaya IP Office from Skype users appear on IP Office phones with no CLID and the Skype username as the Display Name in the following cases:
  - when the Skype user does not have an Online Number
  - when the Skype user does not have a configured Caller ID
- When a Skype user calls a busy extension on IP Office, no busy tone is heard by the Skype user. Avaya IP Offices sends a "486 Busy Here" to Skype, and the Skype call is immediately terminated. The Skype client displays "no answer" for the call. The same result is seen if the Skype user calls an unknown extension on Avaya IP Office (however, IPO sends a "404 Not Found" to Skype, rather than a "486 Busy Here").
- When a Skype user calls a busy extension on IP Office, no busy tone is heard by the Skype user. IPO sends a "486 Busy Here" to Skype, and the Skype call is immediately

terminated. The Skype client displays "no answer" for the call. The same thing happens if the Skype User calls an unknown extension on IP Office (however, IPO sends a "404 Not Found" to Skype, rather than a "486 Busy Here").

- If an IP Office phone calls the PSTN (and the PSTN phone just allows the call to ring), and then the caller abandons the call before 40 seconds there are no issues. However, if caller doesn't abandon the call within 40 seconds, Skype sends a "408 Request Timeout" to Avaya IP Office and the call is terminated.

2. Equipment and Software Validated

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

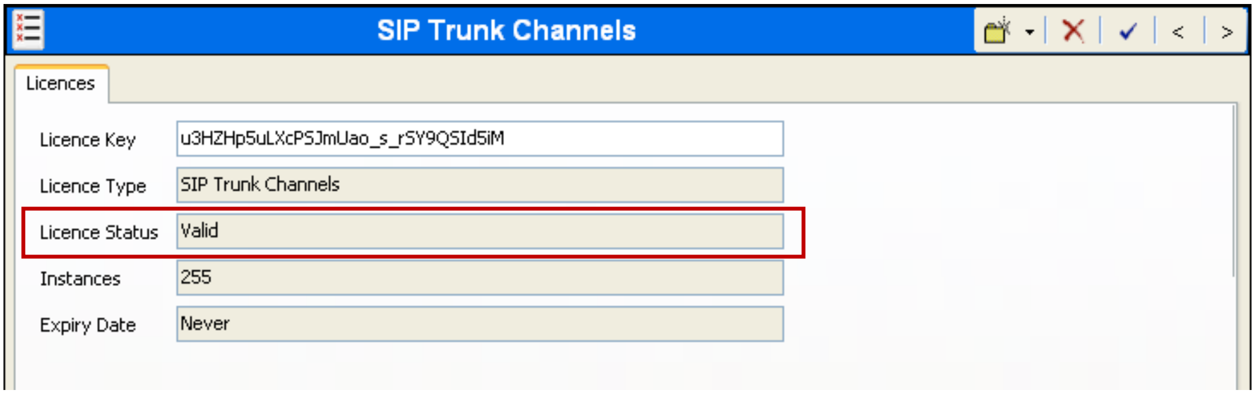
Equipment	Software
Avaya IP Office 500 • DIGSTA8/PRIS U • VCM32 • ANALOG POTS30V2	7.0 (3) • 7.0 (3) • 7.0 (3) • 9.0 (3)
Avaya 2400 Series Digital Telephones	Release 6
Avaya 5600 Series IP Telephones (H.323)	2.9.1
Skype Connect	R2.0

3. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the Skype Connect service. 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 shown in the next section. 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. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the Skype Connect service (such as LAN interface to the enterprise site) is assumed to already be in place.

3.1. Verify Licensing

A license is required for a SIP Trunk from Avaya IP Office to the Skype Connect service. To verify the proper license exists, navigate to **Licence → SIP Trunk Channels** in the Navigation and Group Panes. In the Details Pane, verify the **Licence Status** is ***Valid***.

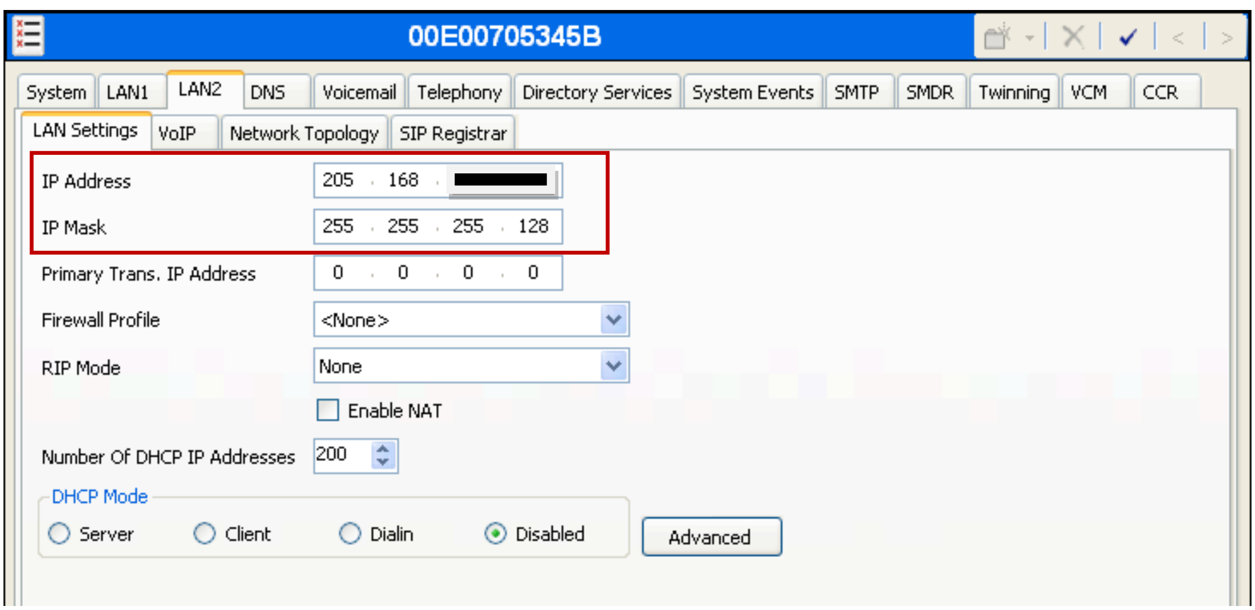


3.2. Configure LAN2 Settings

In the reference configuration, the MAC address ***00E00705345B*** 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.

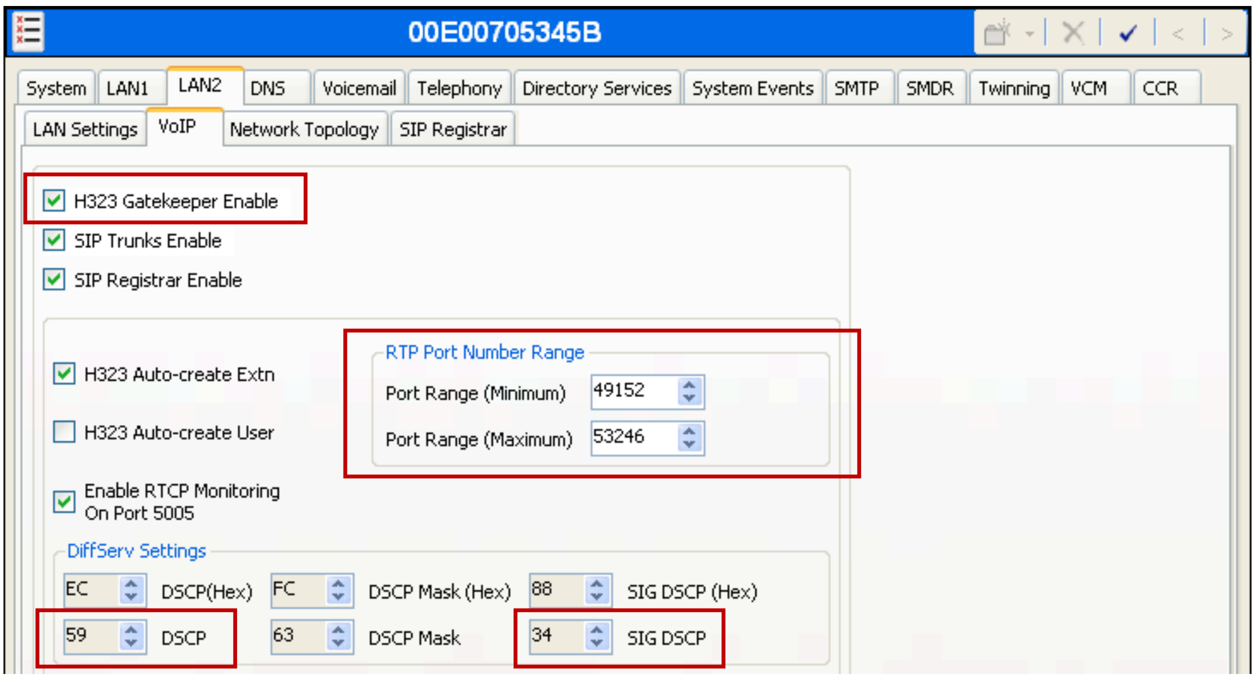
3.2.1. LAN Settings tab

To access the LAN2 settings, first navigate to **System → 00E00705345B** in the Navigation and Group Panes, 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. For security reasons, the second half of the IP address has been hidden in the screen below. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.



3.2.2. VoIP tab

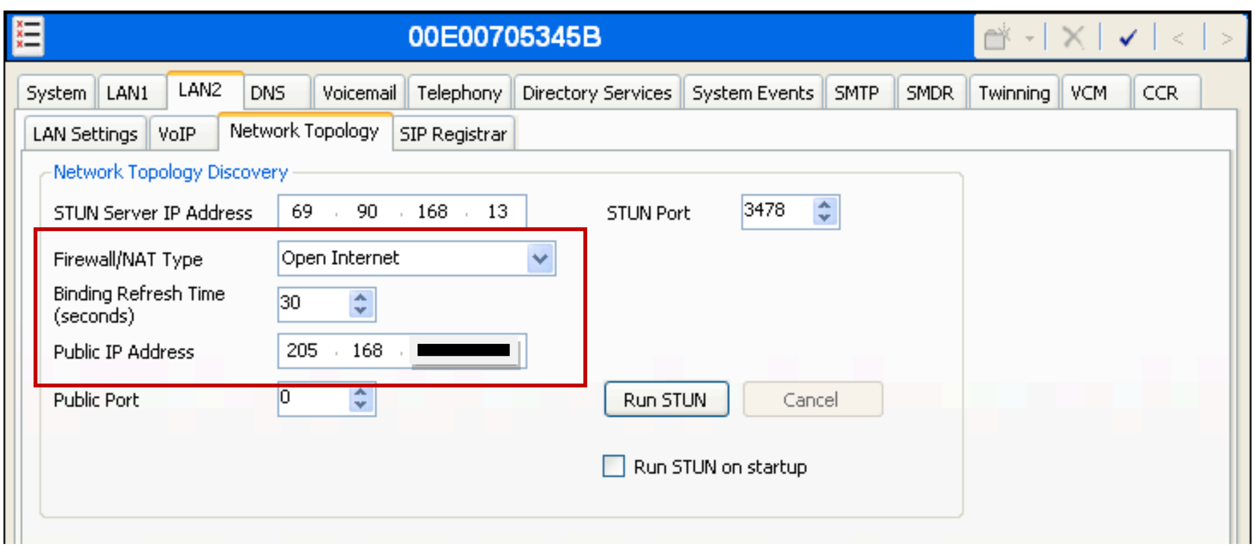
Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Skype. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for during testing are shown in the example below. All other parameters should be set according to customer requirements.



3.2.3. Network Topology tab

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 during testing, as shown in **Figure 1**, so the parameter was set to ***Open Internet***. With this configuration, STUN will not be used.
- Set **Binding Refresh Time (seconds)** to ***30***. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages Skype. Note, the rate at which the SIP OPTIONS are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user (not shown). The SIP OPTIONS period is determined in the following manner:
  - If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used.
  - To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
  - To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port. For security reasons, the second half of the IP address has been hidden in the screen below.
- All other parameters should be set according to customer requirements.



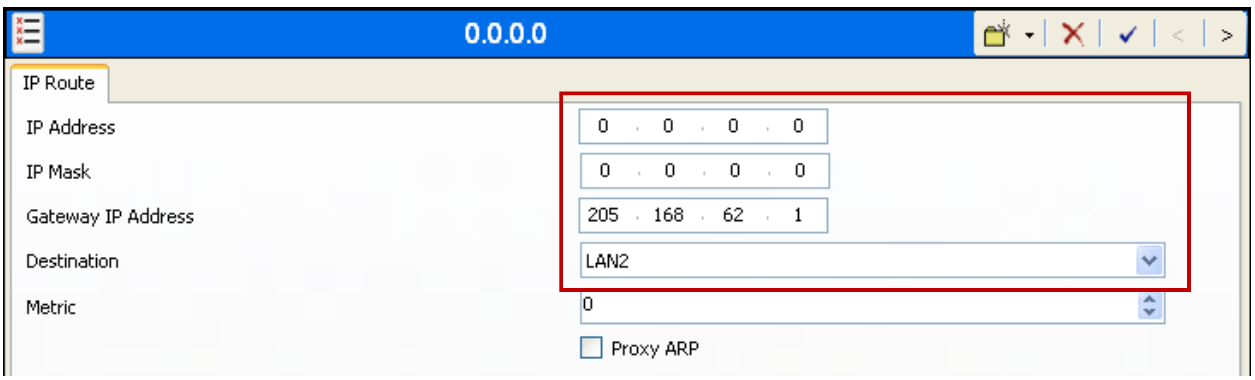
The LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with the Skype Connect service, and therefore it is not described in these Application Notes.

Note, the administrator may select **Run STUN on startup** if the IP Office is behind a NAT/Firewall and IP Office is going to be doing Local NAT compensation. STUN (Simple Traversal of UDP through NAT) is a mechanism used with UDP SIP to overcome the effect of NAT firewalls. The **Run STUN** button is used to test the STUN operation.

3.3. IP Route

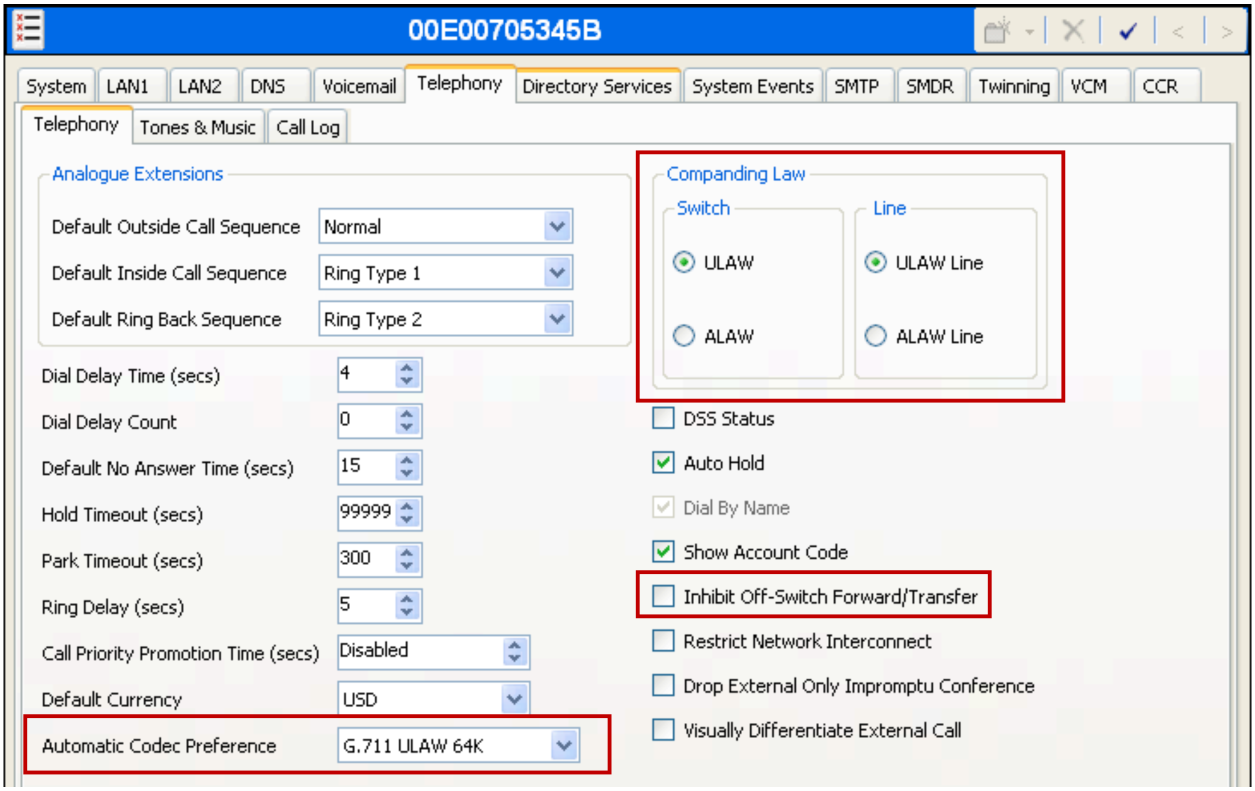
Navigate to **IP Route** in the left Navigation Pane, and then right-click in the Group Pane to select **New**. Create a default route with the following parameters:

- Set **IP Address** and **IP Mask** to ***0.0.0.0***
- Set **Gateway IP Address** to the IP Address of the default router to reach Skype.
- Set **Destination** to ***LAN2*** from the drop-down list.



3.4. System Telephony Settings

Navigate to the **Telephony → Telephony** Tab in 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.



3.5. Administer SIP Line

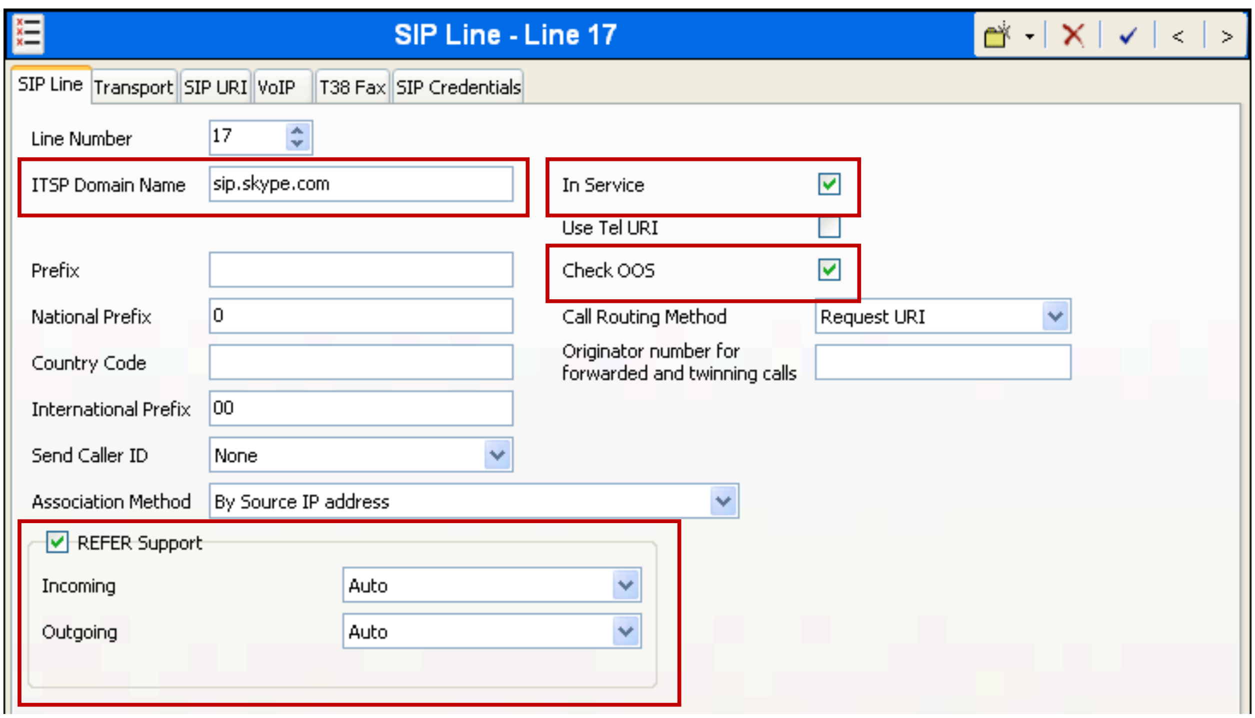
A SIP line is needed to establish the SIP connection between Avaya IP Office and the Skype Connect service. To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line**.

3.5.1. SIP Line tab

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

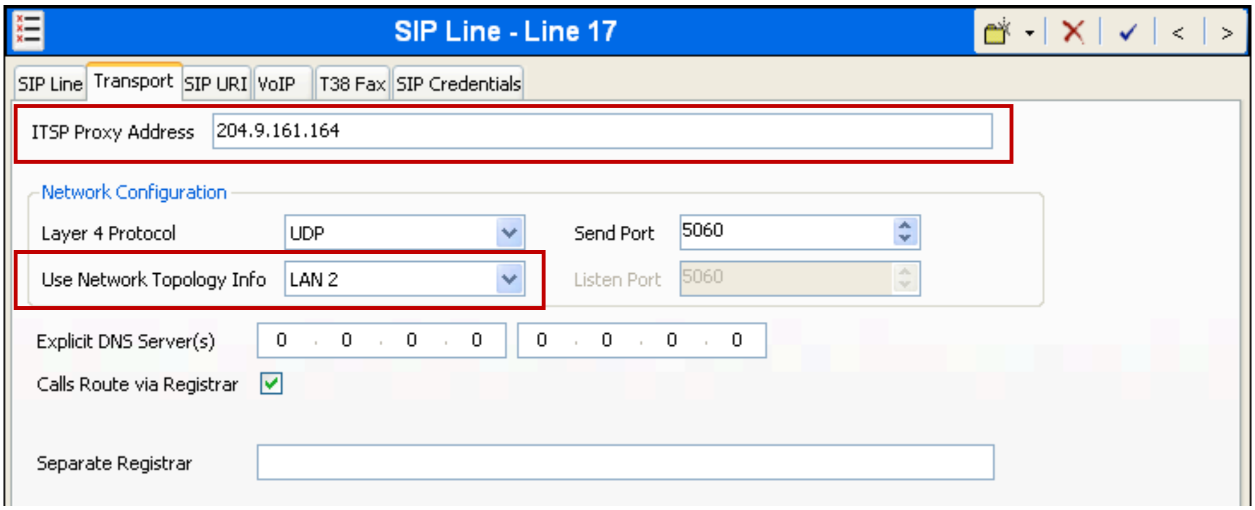
- Set **ITSP Domain Name** to the LAN2 ***sip.skype.com*** so that IP Office uses ***sip.skype.com*** as the host portion of SIP headers such as the From header and Diversion header.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Default values may be used for all other parameters.

In the following screen, the default values of "Auto" are shown for **Incoming** and **Outgoing**, which is sufficient to effectively disable use of SIP REFER for call transfers. To enable SIP REFER, select "Always" from the drop-down menu for **Incoming** and **Outgoing**.



3.5.2. Transport tab

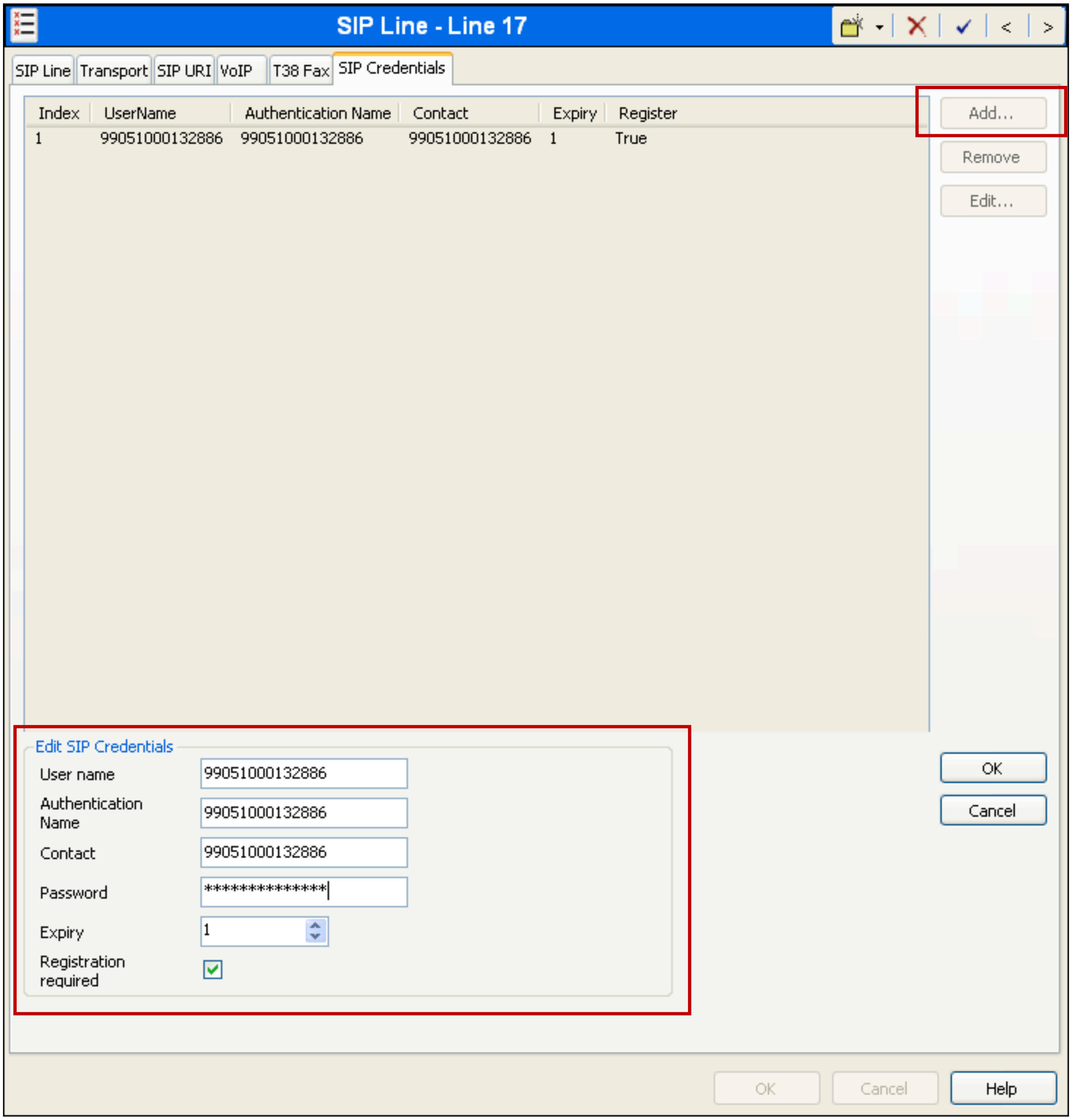
Select the **Transport** tab. The **ITSP Proxy Address** is set to the Skype Connect service IP address provided by Skype. Note, a DNS address may be also be specified for the **ITSP Proxy Address**. If using DNS, the **DNS Service IP Address** field should be populated with the IP address of a DNS Server on the **System→DNS** tab (not shown). As shown in **Figure 1**, the primay IP Address is ***204.9.161.164***. 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 Skype. 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. Other parameters retain default values in the screen below.



3.5.3. SIP Credentials tab

Select the SIP Credentials tab. Click the **Add** button and the **New SIP Credentials** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the bottom of the screen, the Edit SIP Credentials area will be opened. During testing, a single entry was created with the following parameters:

- Set **User name**, **Authentication Name**, and **Contact** to the Skype SIP User that will be used for registration authentication (e.g. 99051000132886).
- Set **Password** to the Skype SIP User password provided for the registration authentication.
- Leave the **Registration required** box checked.



3.5.4. SIP URI tab

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line.

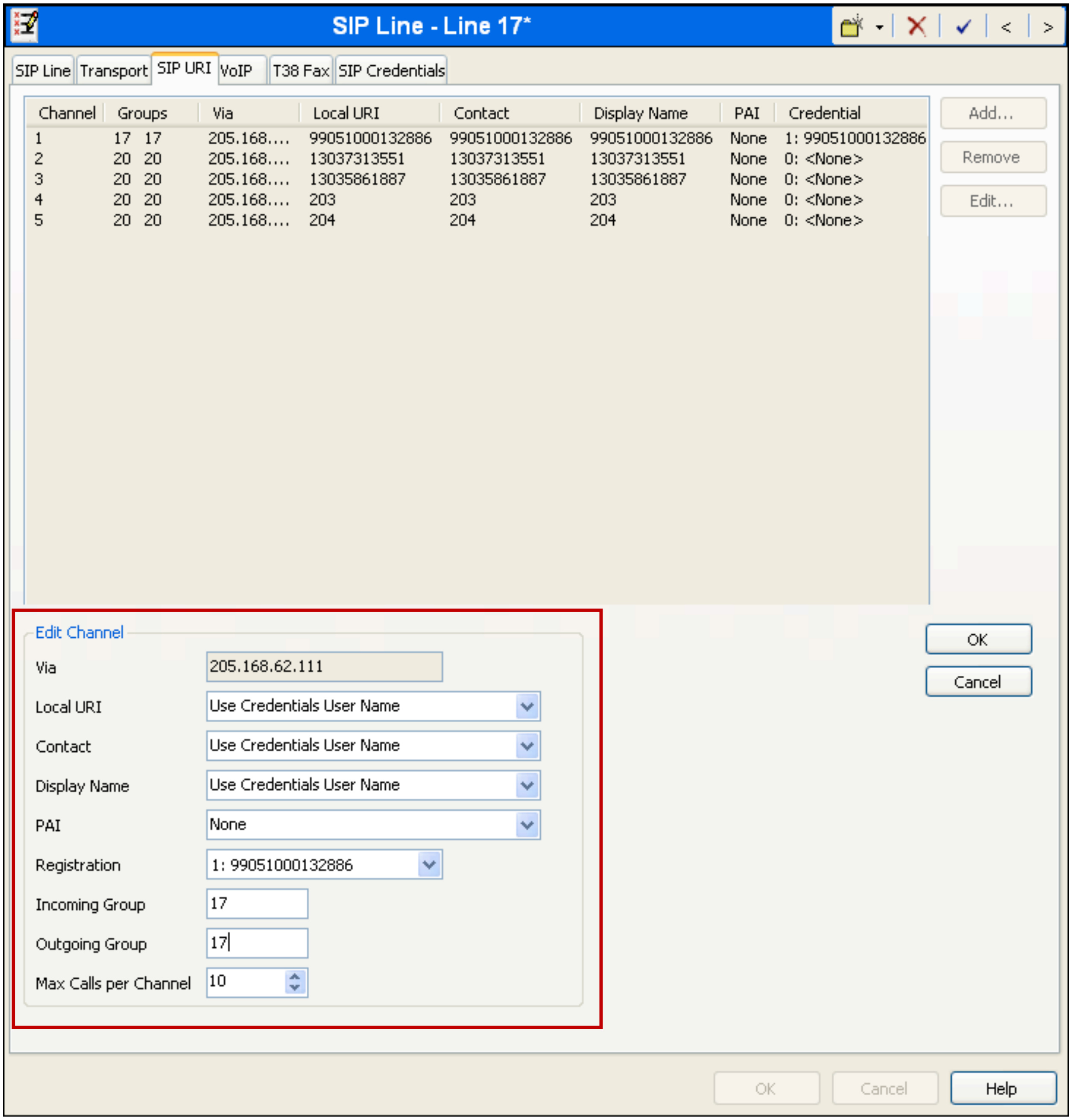
To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the bottom of the screen, the Edit Channel area will be opened. During testing, a single SIP URI entry was for each of the following:

- The Skype SIP User.
- Each incoming call phone number of the Skype SIP User.
- Each Skype business account with an extension number for incoming calls that is associated with the Skype SIP User.

3.5.4.1 Skype SIP User

The SIP URI for the Skype SIP User was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to ***Use Credentials Name***.
- Set **PAI** to ***None***. This field can be used to control whether the PAI should be set, and if so, the source of the value used in the PAI field. This value can either be entered manually or one of the following options may be selected: *None*, *Use Authentication Name*, *Use Credentials User Name*, or *Use Internal Data*.
- The **Registration** parameter is set to the account credentials configured on the line's SIP Credentials tab.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. Enter the number used in **SIP Line – Line Number** which was assigned automatically when the line was created. 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. Enter the same number selected for **Incoming Group**. The outgoing line group number is used in defining short codes for routing outbound traffic to this line.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.



3.5.4.2 Incoming Number

Add an SIP URI for each incoming call phone number of the Skype SIP User.

- Set **Local URI**, **Contact** and **Display Name** to the incoming call phone number of the Skype SIP User (e.g. 13037313551).
- Set **PAI** to ***None***. This field can be used to control whether the PAI should be set, and if so, the source of the value used in the PAI field. This value can either be entered manually or one of the following options may be selected: *None*, *Use Authentication Name*, *Use Credentials User Name*, or *Use Internal Data*.

- Set the **Registration** parameter ***0:<None>***.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. Enter a different number that what was used in **SIP Line – Line Number**.
- Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. Enter the same number selected for **Incoming Group**.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

3.5.4.3 Skype Business Account

Add an SIP URI for each Skype business account with an extension number for incoming calls that is associated with the Skype SIP User.

- Set **Local URI**, **Contact** and **Display Name** to the extension number (e.g. 251) for incoming calls of the Skype business account associated with the Skype SIP User.
- Set **PAI** to ***None***. This field can be used to control whether the PAI should be set, and if so, the source of the value used in the PAI field. This value can either be entered manually or one of the following options may be selected: *None*, *Use Authentication Name*, *Use Credentials User Name*, or *Use Internal Data*.
- Set the **Registration** parameter ***0:<None>***.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. Enter a different number that what was used in **SIP Line – Line Number**.
- Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. Enter the same number selected for **Incoming Group**.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

3.5.5. VoIP tab

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

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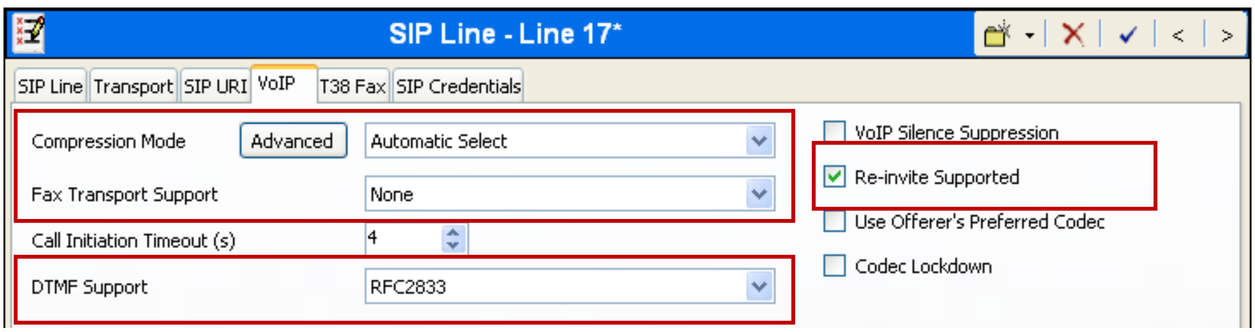
- The **Compression Mode** was configured to ***Automatic Select***. The **Advanced** button can be used to allow an explicit ordered list of codecs to be specified. Check marks next to specific codecs cause Avaya IP Office to include the selected codecs in the Session Description Protocol (SDP) offer, in that order. The Skype supported codec includes:
  - G.711 ALAW 64K
  - G.711 ULAW 64K
  - G.729(a) 8K CS-ACELP
- Set the **DTMF Support** field to ***RFC2833***. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Set the **Fax Transport Support** field to ***None*** since T.38 faxing is not supported by Skype.
- Check the **Re-invite Supported** box.
- Default values may be used for all other parameters.



3.6. Call Routing

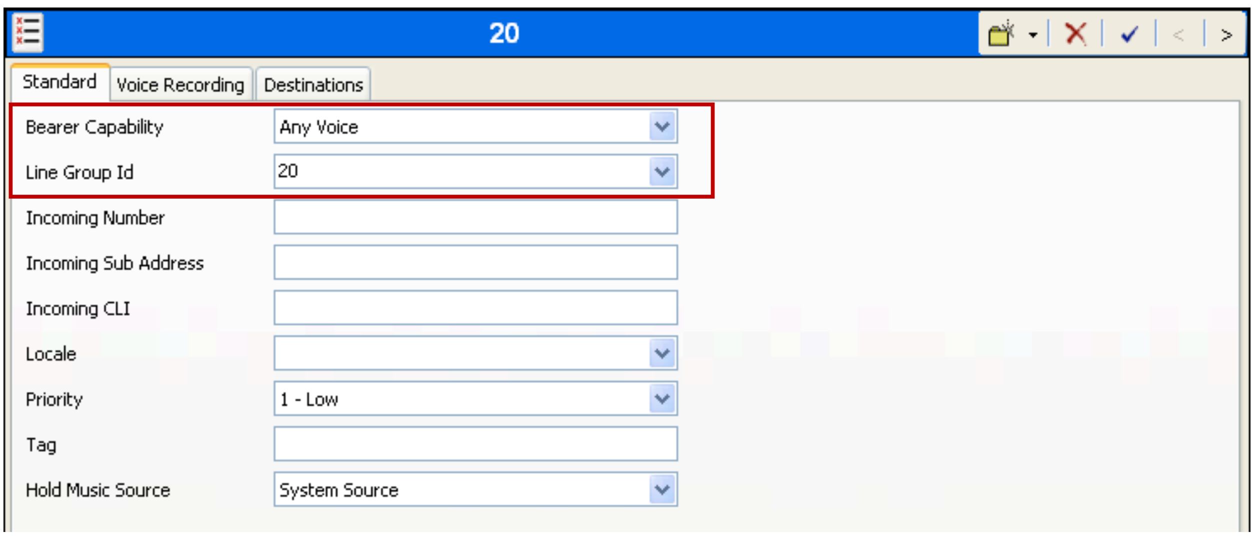
3.6.1. Incoming Call Routing

An incoming call route maps an inbound number on a specific line to an internal extension. This procedure should be repeated for the following:

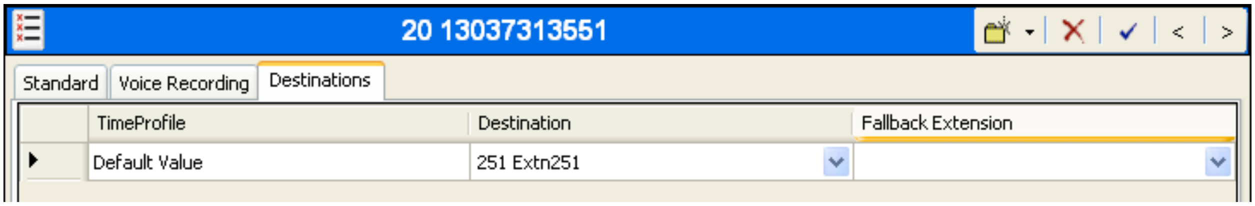
- Each incoming call phone number of the Skype SIP User.
- Each Skype business account with an extension number for incoming calls that is associated with the Skype SIP User.

To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group 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 match the SIP URI – Incoming Group administered in **Section 3.5.4** for each incoming number and business account.
- Default values can be used for all other fields.



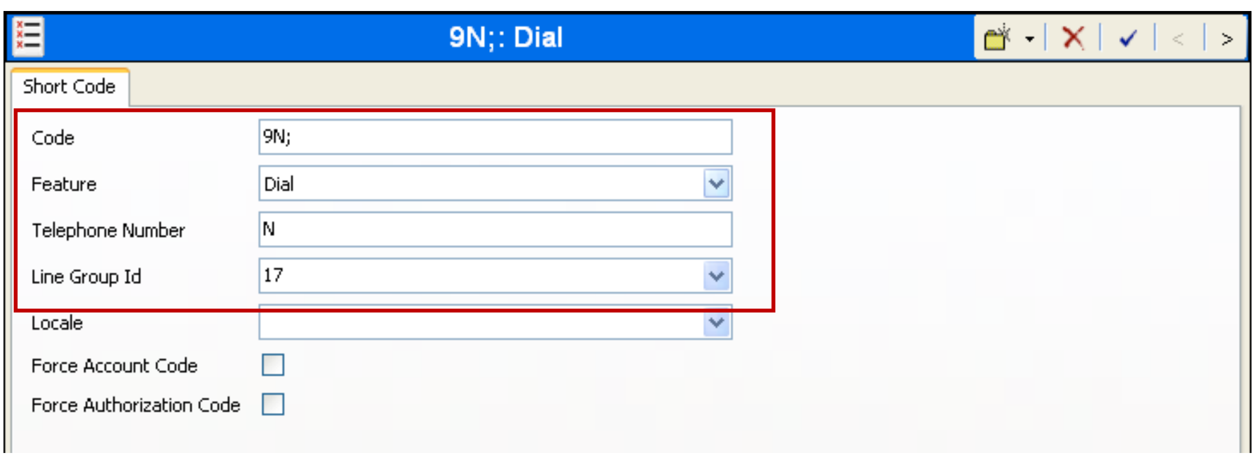
On the **Destinations** tab, either enter the destination manually or select the destination extension from the pull-down menu of the **Destination** field. .



3.6.2. Outgoing Call Routing

Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered "9N;" short code used in the test configuration.

- 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***. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value ***N*** represents the number dialed by the user.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 3.5.4** This short code will use this line group when placing the outbound call.



4. Verification Steps

The correct configuration of the system can be verified by performing the following steps:

1. Verify that the local extensions on Avaya IP Office can call and talk to each other.
2. Verify that the local extensions on Avaya IP Office and the telephones attached to the PSTN can call each other.
3. Verify that if the Skype SIP User has an incoming call phone number that the PSTN can reach the expected Avaya IP Office extension.
4. Verify that if the Skype SIP User has a Skype business account with an extension number for incoming calls that Skype users can call the Skype business account to reach the expected Avaya IP Office extension.
5. Verify that if the Skype SIP User has a Skype business account with no extension number for incoming calls that Skype users can call the Skype business account to reach the expected Avaya IP Office extension.

5. Conclusion

These Application Notes contain instructions for configuring Avaya IP Office to interoperate with the Skype Connect service. All test cases passed with the known limitations noted in **Section 1.2**.

6. Additional References

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

Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

[1] *IP Office Installation,* Document number15-601042, March 2011.
[2] *IP Office Manager,* Document number15-601011, March 2011.
[3] *Voicemail Pro: Installation Manual*, Document Number 15-601063, March 2011.
[4] *System Status Application (IP Office),* Document number15-601758, February 2010.
[5] *IP Office System Monitor*, Document Number 15-601019, November 2008

Product documentation for the Skype Connect service is available from Skype at:
http://www.skype.com/intl/en-us/business/skype-connect

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 Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com.