MaxCS 7.5

SIP-Trunk-Only Orders
SIP Trunk Configuration Guide

Intended audience:

AltiGen Authorized Partners

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Contents

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1. About This Guide

This guide is written for AltiGen Partners who are provisioning SIP Trunks for customers who have ordered SIP Trunks from AltiGen.

These instructions DO NOT apply to customers who have ordered MaxCS Private Cloud Service. Refer to Article 1141 in the AltiGen Knowledgebase for those instructions: https://know.altigen.com/questions/1141/MaxCS+Cloud+Migration+and+SIP+Trunk+Configuration+-+Portal+2.0.

2. Overview

You will follow this general process:

1. After the order has been submitted and processed, open the customer's account in the Cloud Service order portal and switch to the *General* tab. This tab contains configuration details.
2. Purchase SIP Trunk licenses for the premise system, if the customer doesn't have the licenses. Follow the usual process to order these licenses and install them.
3. Configure the SIP trunks. (See section 3 for instructions.)
4. Configure MAXCS as usual.

3. SIP Trunk Configuration

Follow the procedures in this section to configure the SIP Trunks.

3.1. Change Digit 9 to Route Access

SIP Proxy servers typically have different priority values. This indicates that there is a clear order in which the provider would like the servers to be contacted. Calls should route through the primary server, and should only use the secondary server in the event that the primary is not responsive.

In order to accommodate this trunk selection, Out Call Routing must be used on the MaxCS system. This example will use 9 as the leading digit for route selection. If you are using a different digit to get an outside line, adjust accordingly.

1. In MAXCS Administrator, select **System** > **System Configuration**.
2. In the *Trunk Configuration* page, select a SIP Trunk. The *Trunk Access code* field is currently set to 9. Change *Trunk Access code* to **None**. Click **Apply To**.

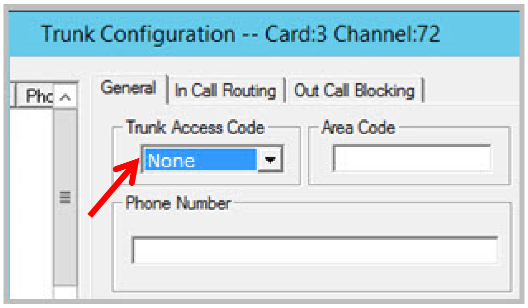


Figure 1: The Trunk Access Code for the SIP Trunk channel

3. All of the other SIP trunks are automatically selected; click **OK**. Click **OK** again to close the window.

4. Select **System** > **System Configuration**. Click the *Number Plan* tab and change 9 from **Trunk Access** to **Route Access**. Click **OK** to close the window.

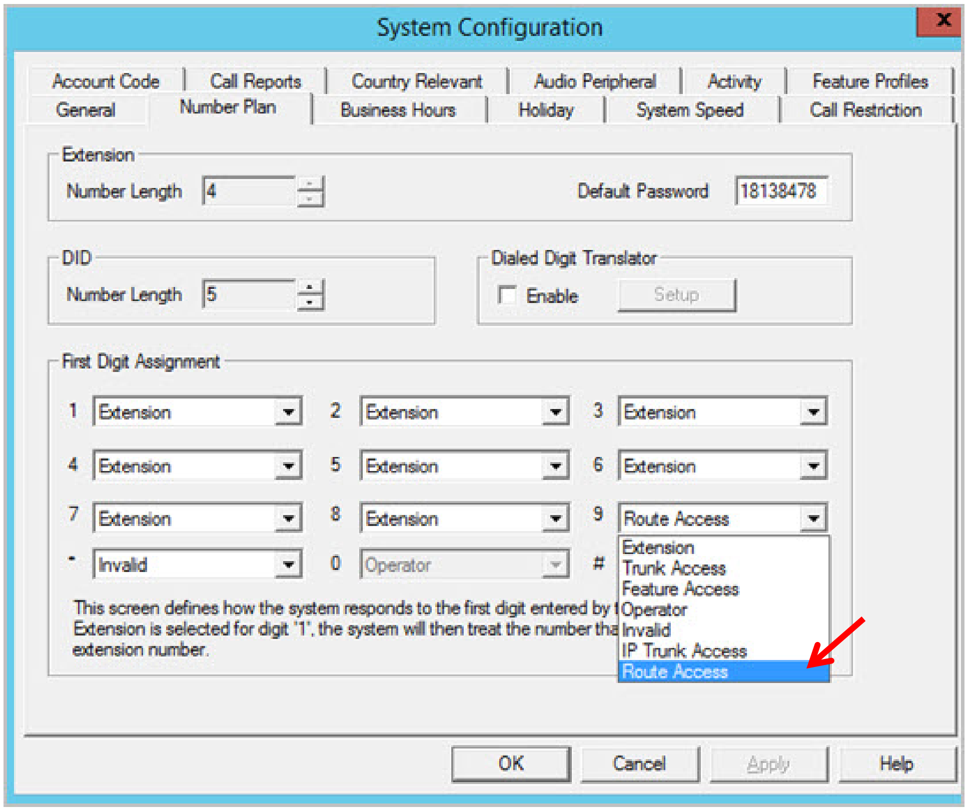
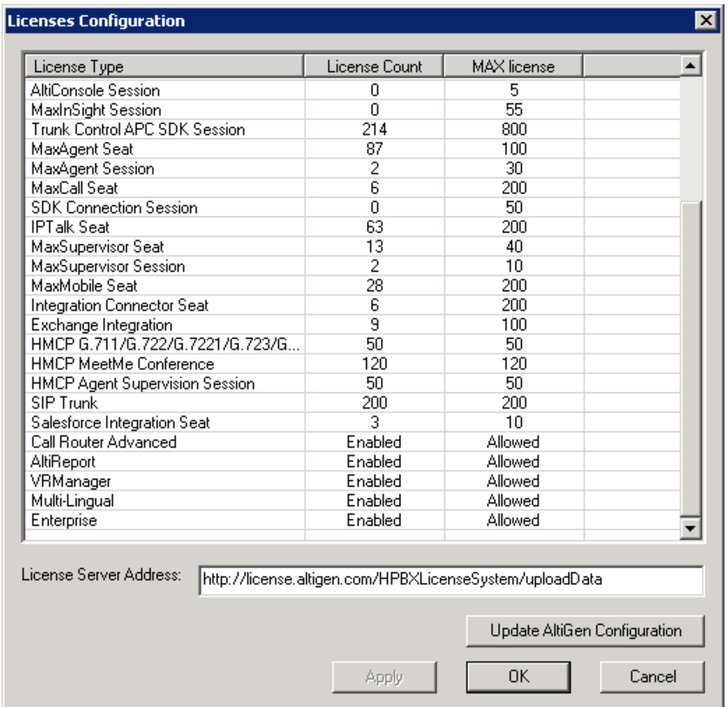


Figure 2: Change Digit 9 to Route Access

3.2. Check Licenses and SIP Trunks

1. Confirm that you have SIP Trunk licenses installed and set for maximum number of channels needed.



Licenses Configuration

License Type	License Count	MAX license
AltiConsole Session	0	5
MaxInSight Session	0	55
Trunk Control APC SDK Session	214	800
MaxAgent Seat	87	100
MaxAgent Session	2	30
MaxCall Seat	6	200
SDK Connection Session	0	50
IPTalk Seat	63	200
MaxSupervisor Seat	13	40
MaxSupervisor Session	2	10
MaxMobile Seat	28	200
Integration Connector Seat	6	200
Exchange Integration	9	100
HMCP G.711/G.722/G.7221/G.723/G...	50	50
HMCP MeetMe Conference	120	120
HMCP Agent Supervision Session	50	50
SIP Trunk	200	200
Salesforce Integration Seat	3	10
Call Router Advanced	Enabled	Allowed
AltiReport	Enabled	Allowed
VRManager	Enabled	Allowed
Multi-Lingual	Enabled	Allowed
Enterprise	Enabled	Allowed

License Server Address: http://license.altigen.com/HPBXLicenseSystem/uploadData

Update AltiGen Configuration

Apply OK Cancel

Figure 3: The License Information panel

2. Make sure that the SIP trunks appear in the *SIP Trunk* view. If they do not appear in that list, you must stop and restart switching before you proceed with SIP trunk configuration.

3.3. Create a New Codec Profile

Next, configure a Codec Profile for the SIP trunks in Enterprise Manager:

1. In MAXCS Administrator, select **VoIP** > **Enterprise Network Manager**.
2. Click the **Codec** button on the *Quick Launch* bar (do not confuse this button with the *IP Codec* tab).
3. Click **Add** and create a profile named *SIPTrunk* with the following parameters:
   - For the name, enter **SIPTrunk.**
   - Set the *Selected Codec* to use **G.711 Mu-Law** (use the **Add** and **Remove** buttons as needed)
   - Set *DTMF Delivery* to **RFC 2833**
   - Set *SIP Early Media* to **Enable**
   - Set *SIP Transport* to **UDP**

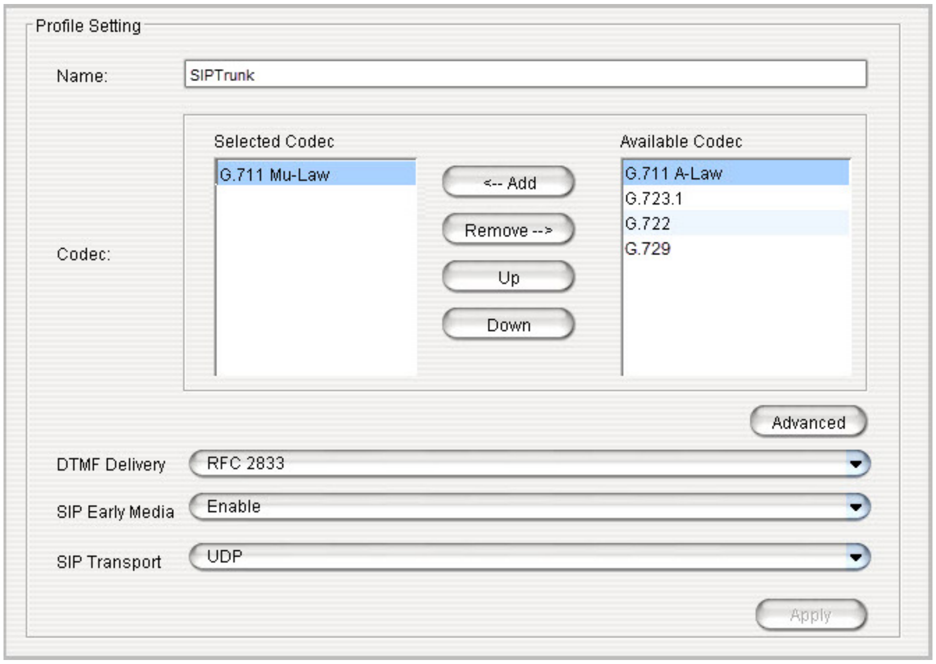


Figure 4: Set SIP Trunk Codec parameters

4. Click **Advanced** and set both packet lengths to **20** ms
5. Click **Ok**.

3.4. Assign the Codec Profile to the Two SIP Servers

Next, assign the *SIPTrunk* Codec Profile to the two SIP trunk servers.

1. In Enterprise Manager, click the **Servers** button on the *Quick Launch* bar.
2. Open the **IP Codec** tab.
3. Configure two servers:
    - 65.254.44.194
    - 74.81.71.18

    a) Click **Add.**

    b) Add the first server (65.254.44.194) and set the *Codec Profile* to **SIPTrunk.**

    c) Repeat these steps for the second server.

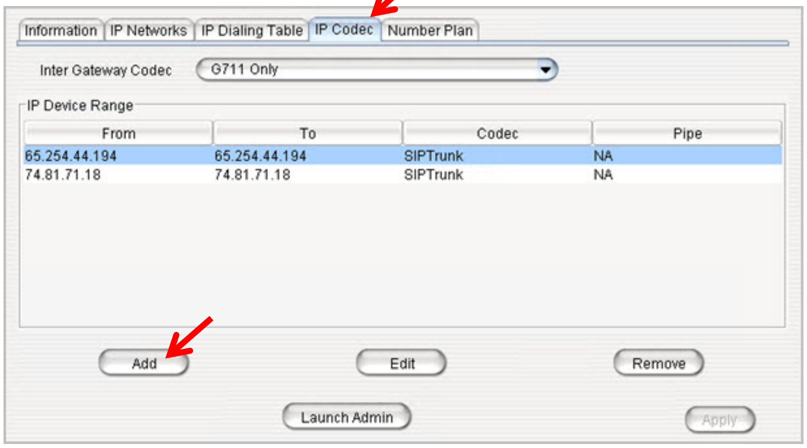


Figure 5: The Servers page (IP Codec tab) in Enterprise Manager

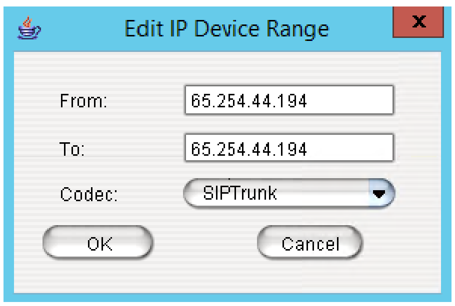


Figure 6: Server assigned to the Codec Profile SIPTrunk

3.5. NAT Configuration

1. Click the **IP Networks** tab and check both of the NAT support checkboxes. Confirm that the *Public IP Address* matches the public IP address of this system.

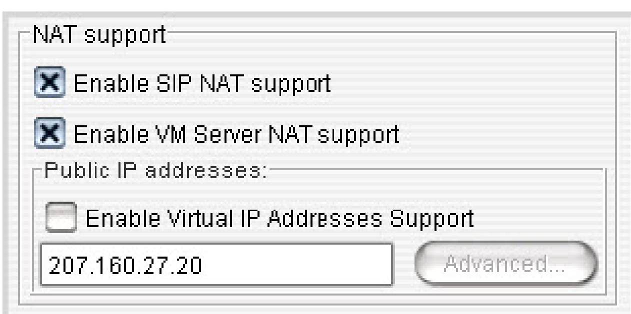


Figure 7: Enable NAT Support

2. Close Enterprise Manager.

3.6. Create a New SIP Trunk Profile

1. Open the SIP Trunk Configuration panel:
    a. In MAXCS Administrator, open Trunk view.
    b. Double-click an unconfigured SIP Trunk.
    c. In the *Trunk Configuration* panel, click **Trunk Properties**.
    d. In the next dialog box, click **SIP Trunk Configuration**.
    e. Click **SIP Trunk Profile**.

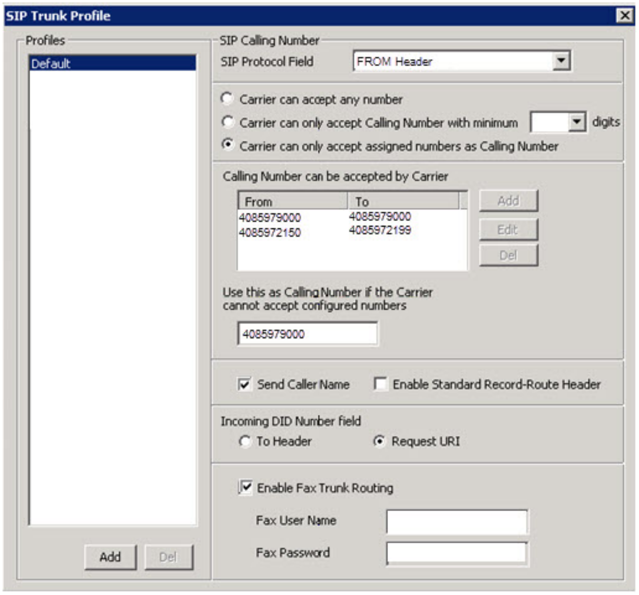


Figure 8: The SIP Trunk Profile panel

2. Click **Add** and give the new profile the name **SIPTrunk**.
3. Configure the parameters as follows:
    - Set *SIP Protocol Field* to **FROM Header**.
    - Select **Carrier can only accept assigned numbers as Calling Number**.
    - Enter the range of DID numbers that you want to assign to this trunk in the *Calling Number can be accepted by the Carrier* box. These must be valid DIDs that are on the trunk; each entry must be 10 digits.
    - For the *Use this Calling Number if the Carrier cannot accept configured numbers* field, enter the main phone number. The phone number that you enter must be included among the *Calling Number can be accepted by the Carrier* entries.
    - Select **Send Caller Name**. (Do NOT select Enable Standard Record-Route Header.)
    - Set *Incoming DID Number field* to **Request URI**.
4. Click **OK**.

3.7. Plan the Channel Configuration

You will need the SIP Trunk details from your order in the MAXCS Cloud portal to complete the steps in this section.

Note: If you are configuring FoIP, reserve the **last two available channels** for FoIP configuration. See page 18 for details.

For failover purposes, you will assign half of the channels to the first server, 65.254.44.194, and the remaining half to the second server, 74.81.71.18.

In addition, you will configure only the **first channel** in each half to register.

1. Determine the number of SIP Trunk channels available, and determine the channel number mid-way through the list. This channel will be the first of the channels that uses the second server.

   For example, if you have 20 channels, consider the 11th channel as the channel that switches to the second server. You would assign the first 10 channels to the first server, and the last 10 channels to the second sever. (See Figure 9 for an example.)

2. Jot down this mid-point channel number; we'll refer to this as ChannelB in later steps, to avoid confusion. You can also note the range of channels for the first and second servers.

Channel number	Server	Register period
0072	01	60
0073	01	0
0074	01	0
0075	01	0
0076	01	0
0077	01	0
0078	01	0
0079	01	0
0080	01	0
0081	01	0
0082	02	60
0083	02	0
0084	02	0
0085	02	0
0086	02	0
0087	02	0
0088	02	0
0089	02	0
0090	02	0
0091	02	0

ChannelB

ChannelB: ____________

1st Server channels: _________ - ________

2nd Server channels: ________ - ________

Figure 9: Example of dividing 20 channels between two servers

3.8. Collect the Configuration Details

1. Log into the MAXCS Cloud Order portal.
2. Open the order for which you are configuring the SIP services. Click the *General* tab.

This tab, in the *Voice SIP Trunk Group* section under *SIP Services*, contains the SIP Trunk details.

You can copy and paste the information from the order portal into MAXCS Administrator, so leave this browser window open for now.

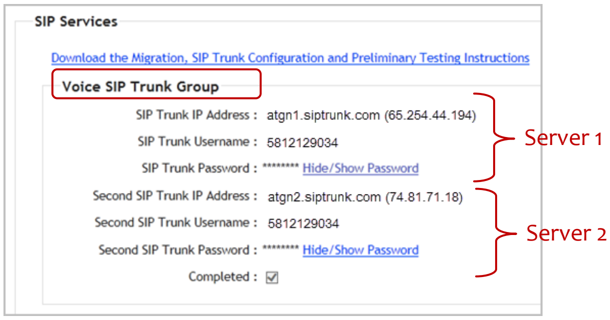


Figure 10: The Voice SIP Trunk Group information in the Cloud Order portal

3.9. Configure the First Half of the Channels

Configure the first half of the SIP Trunk channels, which will be assigned to the *first* server.

1. Switch to MAXCS Administrator. You should be in the *SIP Trunk Configuration* page. If not, open it now.
2. Select the first SIP Trunk channel and click **Edit**. Set the following parameters and click **OK**:
   - For *SIP Server IP Address*, enter the IP address of the first SIP trunk server (in our example, this is 65.254.44.194).
   - For *User Name*, paste the *SIP Trunk Username* number from the order in the Cloud portal. In Figure 9, for example, this username is *5812129034*.
   - For *Password*, paste the *SIP Trunk Password* string from the order in the Cloud portal. Click **Hide/Show Password** to see it.
   - For *Domain*, paste the URL from the order in the Cloud portal: *atgn1.siptrunk.com*.
   - Set the *SIP Register Period* to **0**. (You will modify this for the first channel later)
   - Set the *SIP Trunk Profile* to **Default**.
   - Set the *SIP Source Port* to **5060**.
   - Set the *SIP Destination Port* to **5060**.
   - Select **Enable Channel**.

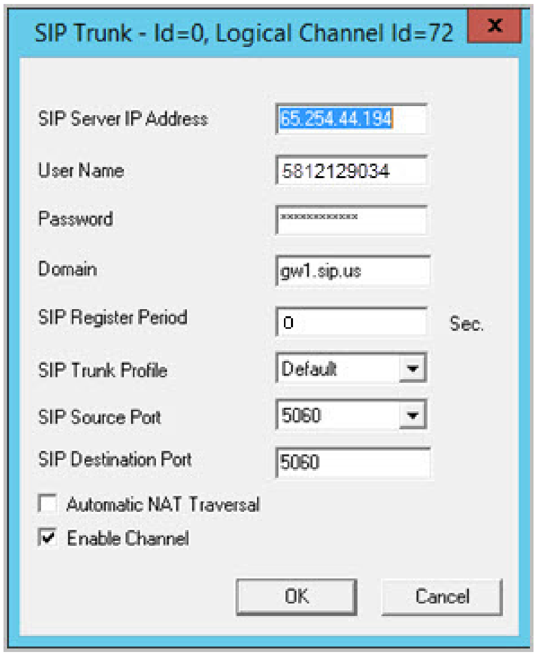


Figure 11: The configuration for the first channel

3. Copy that configuration to the rest of the ***first half*** of the channels:

    a. Select the first channel and click **Copy To.**

    b. Select the range of channels from the second channel through the end of the channels that belong with the first server. Do not include ChannelB or any channels beyond ChannelB. For the scenario in Figure 9, for example, you would copy the configuration of channel 72 to channels 73 through 81.

    c. Click **OK**.

4. Update the first channel to specify a registration period other than 0, so that this trunk will send a registration message to the carrier:

    a. Select the first SIP Trunk channel again. Click **Edit**.

    b. Change the *SIP Register Period* setting to **60.** Click **OK**.

3.10. Configure the Second Half of the Channels

Now you will configure the **second half** of the channels, which will be assigned to the *second* server. This process is almost identical to step 3.9, except that you are using the "Secondary" SIP details from the order portal instead of the details for the first sever.

1. Select the first channel of the second half (ChannelB). Click **Edit**.

2. Set the following parameters and then click **OK**.

    - For *SIP Server IP Address*, enter the IP address of the second SIP trunk server (in our example, this is 74.81.71.18). This address will be different from the IP address you entered for the first set of channels.

- For *User Name*, paste the same string as for the first set of channels. In Figure 9, for example, this is *5812129034*.
- For *Password*, paste the same string as for the first set of channels; click **Hide/Show Password** to see it.
- For *Domain*, paste the *URL* from the *second* SIP Trunk address in the Cloud portal. This will be different from the URL for the first set of channels. In Figure 9, for example, this is *atgn2.siptrunk.com*.
- Set the *SIP Register Period* to **0**. (You will modify this again, for just ChannelB, later)
- Set the *SIP Trunk Profile* to **Default**.
- Set the *SIP Source Port* to **5060**.
- Set the *SIP Destination Port* to **5060**.
- Select **Enable Channel**.

3. Copy the configuration of ChannelB to the rest of the ***last half*** of the channels:

    a. Select ChannelB and click **Copy To.**

    b. Select the range of channels from the next channel (after ChannelB) through all of the rest of the channels. In Figure 9, for example, you would copy the configuration of channel 82 to channels 83 through 91.

    c. Click **OK**.

4. Now edit ChannelB to specify a registration period other than 0, so that this trunk will send a registration message to the carrier:

    a. Select ChannelB. Click **Edit**.

    b. Change the *SIP Register Period* setting to **60.** Click **OK**.

3.11. Check Your Configuration

1. In the SIP Trunk Configuration panel, the first half of your channels should list the first server and the remaining channels should list the second sever.

    In addition, the first channel and ChannelB should both show a registration period of 60 seconds; all other channels should show that field as empty.

    In Figure 12, ChannelB is channel number 122 (ID 50); channels above it are assigned the first server, the rest are assigned to the second server, and ChannelB has a Register Period of 60 seconds.

2. If your configuration is not correct, make the necessary adjustments before you continue.

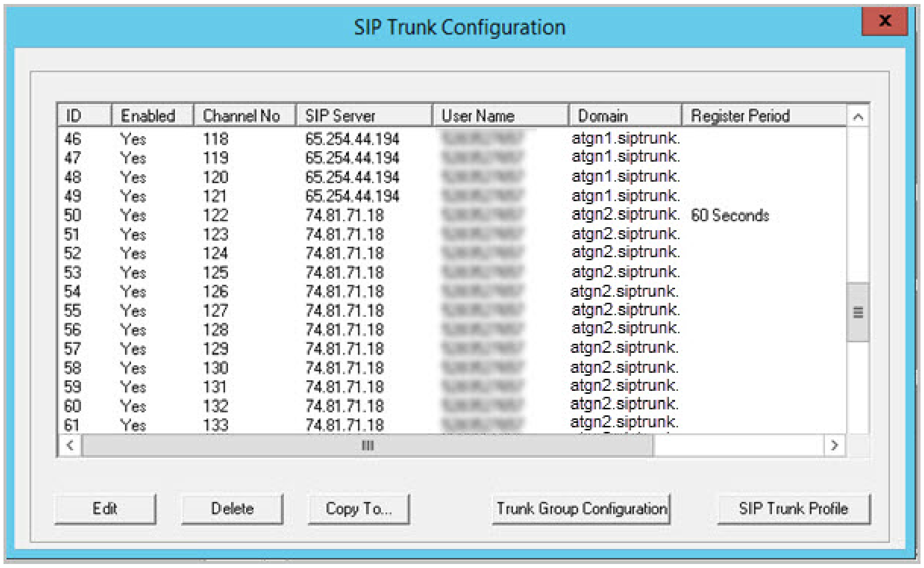


Figure 12: Channels are assigned to the second server starting with channel number 122 (which is ID 50)

3. Return to the main Trunk view (close the various panels to reach that view) and confirm that all configured SIP trunk channels show as *Idle*, including the two channels that are configured to register.

3.12. Enable SIP Options for the Servers

The next step is to configure Enable SIP options for both of these servers. By sending a "keepalive" message and checking for a valid response, SIP devices will know whether remote peers are ready to receive a new request.

1. You should still be on the *SIP Trunk Configuration* page. Click **Trunk Group Configuration**.
2. Select the first SIP Trunk server (refer to the example IP addresses in section 3.4 on page 6) and assign it the name *SIPTrunk_Primary*. Check the **Enable SIP OPTIONS** checkbox and click **Apply**.
3. Select the second SIP Trunk server (refer to the example IP addresses in section 3.4 on page 6) and assign it the name *SIPTrunk_Secondary*. Check the **Enable SIP OPTIONS** checkbox and click **Apply**.

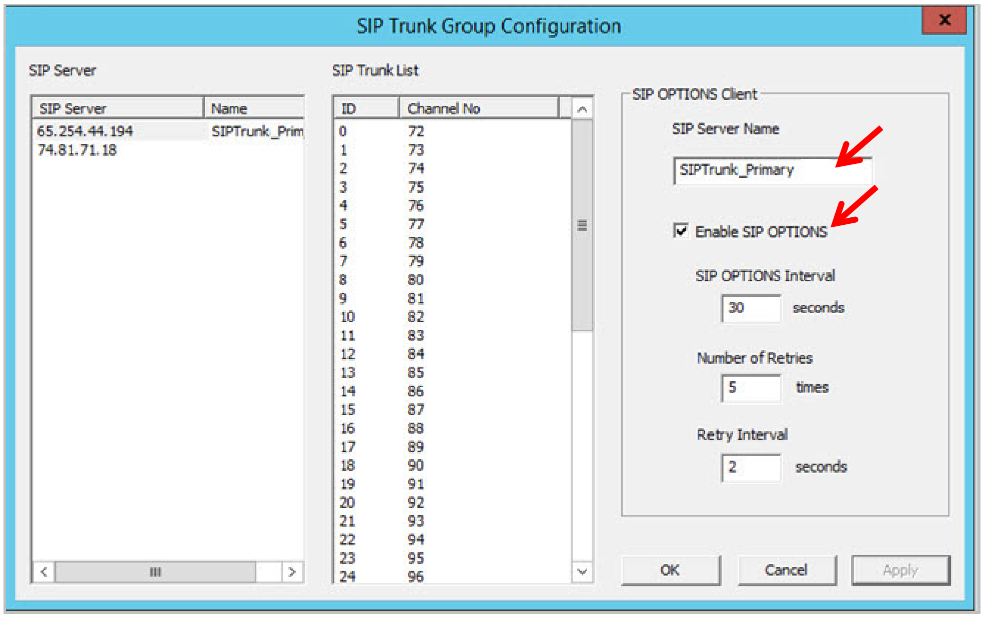


Figure 13: Enable SIP OPTIONS for both servers

4. Close the various panels until you return to Trunk View. The first channel and ChannelB should both be green with a status of *Idle*. If one of these channels does not show as Idle, check your configuration.

3.13. Configure Out Call Routing

Finally, configure out-call routing rules.

1. Identify the system home area code: Select **System** > **System Configuration**. Switch to the *General* tab and note the *System Home Area Code*.
2. Select **PBX** > **Out Call Routing Configuration**. On the *Route Definition* tab, add a route called *Primary*:
   a. Click **Add**. Leave the index number as is and enter the name *Primary.*
   b. To add member trunks, select the channels that are assigned to the first server in the *Not Member* list (refer back to page 9, where you identified ChannelB). Click the **Left Arrow** button to add these channels to the *Member Trunks* list. Click **Apply**.
3. Add three additional routes:
   a. Follow the same process as the preceding step to add another route, called *Secondary*. This time, add the channels that are assigned to the *second* server. Click **Apply**.
   b. Create a third route, called *Primary Local*. Add the channels for the first server. In addition, under *Digit Manipulation*, check the box and select **Insert to Head**. Add the system area code you identified in step 1, preceded by the numeral 1. Click **Apply**.
   c. Add a fourth route, *Secondary Local*. Add the channels for the second server. Check the box and select **Insert to Head**. Add the system area code preceded by the numeral 1. Click **Apply**.

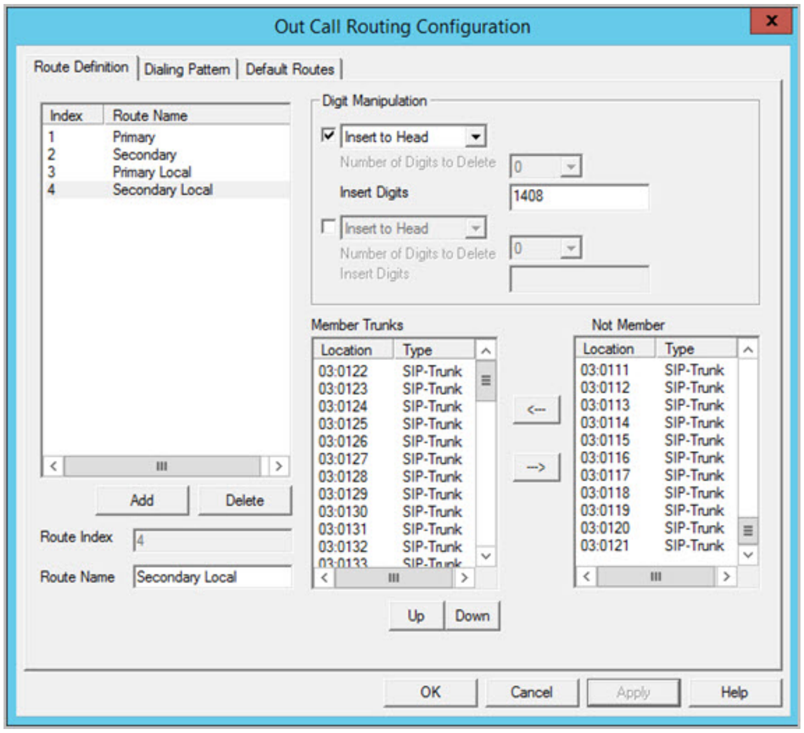


Figure 14: Out Call Routing configurations

4. Switch to the *Default Routes* tab. Set the following routes and click **OK**.

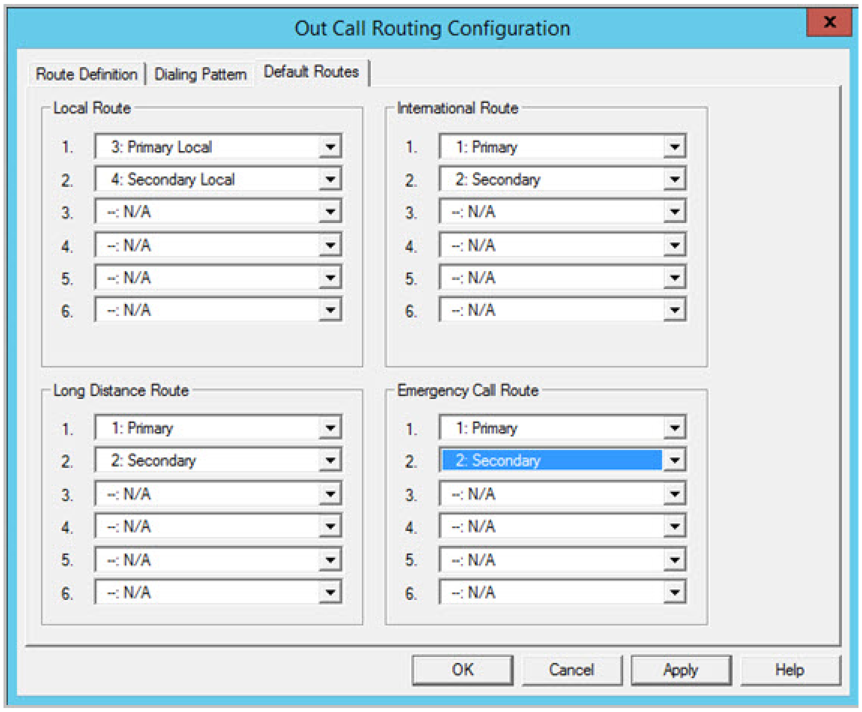


Figure 15: Default route configuration

- Set *Local Route 1* to **3: Primary Local**
- Set *Local Route 2* to **4: Secondary Local**
- Set *International Route 1* to **1: Primary**
- Set *International Route 2* to **2: Secondary**
- Set *Long Distance Route 1* to **1: Primary**
- Set *Long Distance Route 2* to **2: Secondary**
- Set *Emergency Call Route 1* to **1: Primary**
- Set *Emergency Call Route 2* to **2: Secondary**

3.14. Configure Inbound Routing

AltiGen SIP trunks receive 11 digits as DNIS on inbound calls; configure your inbound routing rules accordingly.

3.15. Configure FoIP (Optional)

For Fax-over-IP (FoIP) configuration, refer to the MAXCS 7.5 Administration Manual.

4. SIP Trunk Testing

These tests should be performed against the SIP trunk once all configuration steps have been completed. You can print the last few pages to make notes as you test.

4.1. Confirm the SIP Trunk IP Address

Confirm that the correct IP address has been entered in Enterprise Manager; open the **Servers** tab and switch to the **IP Codec** sub tab. Check that the correct IP address is listed in the *IP Device Range* table.

4.2. Test Inbound Calls to Test Extensions

While performing these tests, note that the carrier sends 11 digits as DNIS.

1. Create a test extension; include the test DID (if applicable) in the extension's DID field.
2. Register the IP Phone using the new test extension.
3. Test the following:
    a. Place an inbound call to the test extension.
    b. Place the call on hold.
    c. Transfer the call to an external number.

DIDs to be tested inbound (test 5 numbers):

1. Number: ________________ Dest. (AA, Ext, WG): ________________ Pass/Fail: _____ Notes: ________________
2. Number: ________________ Dest. (AA, Ext, WG): ________________ Pass/Fail: _____ Notes: ________________

3. Number: ______________ Dest. (AA, Ext, WG): ______________ Pass/Fail: _____ Notes: ______________

4. Number: ______________ Dest. (AA, Ext, WG): ______________ Pass/Fail: _____ Notes: ______________

5. Number: ______________ Dest. (AA, Ext, WG): ______________ Pass/Fail: _____ Notes: ______________

4.3. Test Outbound Calls to External Numbers (PSTN)

1. Place an outbound call to a PSTN number.
2. Verify that the outbound Caller-ID is transmitted.
3. Set call restrictions and test.

---

1. Ext Type (IP Phone/Analog phone/Fax): ______________ External Type (Toll free, LD, Local): __________

   Pass/Fail: _____ Notes: ____

2. Ext Type (IP Phone/Analog phone/Fax): ______________ External Type (Toll free, LD, Local): __________

   Pass/Fail: _____ Notes: ____

3. Ext Type (IP Phone/Analog phone/Fax): ______________ External Type (Toll free, LD, Local): __________

   Pass/Fail: _____ Notes: ____

4. Ext Type (IP Phone/Analog phone/Fax): ______________ External Type (Toll free, LD, Local): __________

   Pass/Fail: _____ Notes: ____

5. Ext Type (IP Phone/Analog phone/Fax): ______________ External Type (Toll free, LD, Local): __________

   Pass/Fail: _____ Notes: ____

4.4. Test Extension to Extension

1. Configure a 2nd test extension.
2. Test the following:
   a. Place a call to a 2nd test extension.
   b. Place the call on hold.
   c. Place a call from an external number to the test DID.
   d. Have one of the test extensions answer.
   e. Once answered, transfer the call to the other test extension.

Pass/Fail: _____ Notes: ______________________________________

If you have any technical issues during the testing, please contact AltiGen Technical Support at 1-888-ALTIGEN (258-4436) Option 5.

5. FoIP Configuration

This section describes how to configure FoIP; you will also need the article on your specific Fax device; find these articles in the AltiGen Knowledgebase.

5.1. Edit the SIP Trunk Profile to Enable Fax Routing

You can allow voice and fax calls to run on the same SIP trunk channel. The trunks must be configured to support both voice and fax. The SIP trunk uses the same SIP server IP address, but different authentication credentials for voice trunk versus a fax trunk.

To update the SIP Trunk profile that you created earlier, follow this process.

1. In MAXCS Administrator, open **Trunk view**.
2. Double-click an unconfigured SIP Trunk.
3. In the Trunk Configuration panel, click **Trunk Properties**.
4. In the next dialog box, click **SIP Trunk Configuration**.
5. Click **SIP Trunk Profile**.
6. Select the **SIPTrunk** profile.
7. Check **Enable Fax Trunk Routing**.

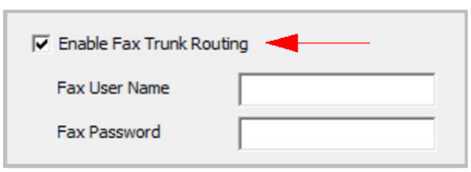


Figure 16: Enable Fax Routing

8. Enter the fax trunk's user name and password (refer to Figure 17). Click **OK**.

Outbound calls that are made through SIP channels that have been configured for fax channels are for fax only. Therefore, they should **not** be assigned trunk access codes or be included in the out call routing for voice calls.

For AltiGen SIP trunks, you must configure one SIP trunk channel to perform SIP registration for ATGN1.SIPTRUNK.com and ATGN2.SIPTRUNK.com of the fax trunk, as described in the next section.

5.2. Configure Two Fax Registration Channels

You will need to retrieve the fax username and password from the order in the Cloud Services portal in order to complete these procedures.

To configure two fax channels to register to the two AltiGen SIP gateways,

1. Log into the MAXCS Cloud order portal and retrieve the Fax details on the *General* tab of your account.

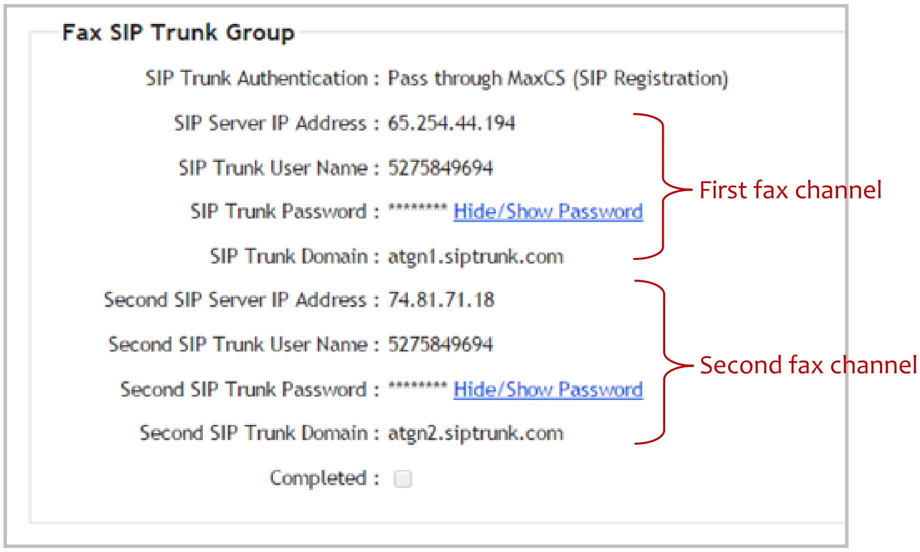


Figure 17: Fax configuration details in the order portal

2. Back in MaxAdministrator, on the *SIP Trunk Configuration* page, select a channel to configure and click **Edit**.
3. Configure this channel with the following parameters, and click **OK**.
   - For *SIP Server IP Address*, enter the IP address of the first SIP trunk server, **65.254.44.194**
   - For *User Name*, enter the fax username (in the example in Figure 17, this is 5275849694)
   - For *Password*, enter the SIP Trunk Password (click **Hide/Show Password** to see it)
   - For *SIP Trunk Domain*, paste the URL from the first SIP Trunk address in the Cloud portal. In the example, this is *atgn1.siptrunk.com*
   - Set the *SIP Register Period* to **60**
   - Set the *SIP Trunk Profile* to **Default**
   - Set the *SIP Source Port* to **5060**
   - Set the *SIP Destination Port* to **5060**
   - Select **Enable Channel**
4. Configure a second channel with the following parameters, and click **OK**.
   - For *SIP Server IP Address*, enter the IP address of the second SIP trunk server, **74.81.71.18**
   - For *User Name*, enter the fax username (in the example in Figure 17Figure 17: Fax configuration details in the order portal, this is 5275849694)
   - For *Password*, enter the SIP Trunk Password
   - For *Domain*, paste the URL from the second SIP Trunk address in the Cloud portal. In our example, this is *atgn2.siptrunk.com*
   - Set the *SIP Register Period* to **60**
   - Set the *SIP Trunk Profile* to **Default**
   - Set the *SIP Source Port* to **5060**
   - Set the *SIP Destination Port* to **5060**
   - Select **Enable Channel**

Do not add these two SIP Trunk channels to any out call routing tables.

Do not assign trunk access codes to these two fax SIP Trunk channels.

These two fax SIP Trunk channels can receive either voice or fax trunk calls.

5.3. Configure the Fax Device

MAXCS works with several Fax devices. For instructions on configuring your device, search the AltiGen Knowledgebase for articles on your device's manufacturer and model.