# MAXCS Release 7.5

## All-Software Solution
## Deployment Guide

**Intended audience:**
**AltiGen Authorized Partners**

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# Introduction

This document is provided for AltiGen partners who will be performing a fresh installation of MAX Communication Server (MAXCS) ACM Release 7.5 All-Software Softswitch Solution for their clients.

There are two types of Softswitch deployments for MaxCS 7.5:

- **Small Business Softswitch**
- **Standard Softswitch**

Some notes regarding this Softswitch installation:

- This is a software-only installation process.
- This installation does not support redundancy or gateway installation.
- All components must be installed on the same server.

## The Softswitch Small Business Deployment

The Softswitch Small Business system is capped at 24 users and cannot be expanded. It includes the following licenses:

- 1 ACM Base license
- 1 HMCP Media license
- 24 Combo Station licenses
- 24 SIP Trunk licenses
- 24 Compressed VoIP Channel licenses
- 12 Meet Me Conference Resources

## The Softswitch Standard Deployment

The Softswitch Standard bundle includes the following licenses:

- 1 ACM Base license
- 30 Port MeetMe Conference
- HMCP Media Server
- 1 AltiConsole license
- 1 Multilingual license
- 1 AltiEnterprise License
- 5 HMCP Combo Codecs
- 5 Agent Supervision

**Note:** These licenses can only be registered with the Softswitch system key.

## Additional Licenses Available

The following additional license bundles are available for the Softswitch Standard deployment:

- The **Contact Center Combo license** bundle (ALTI-ACMCOMBO) includes the following licenses:
  - 5 Agent Combo licenses
  - 1 Supervisor Combo license
  - 1 AltiReport license
  - 1 MaxInsight license
  - 1 Advanced Call Router license
- The **Station Combo license** bundle (ALTI-COMBOSTATION-XX) includes the following licenses:
  - 1 Softswitch Station license
  - 1 MaxCommunicator/AltiView license
  - 1 IP Talk license
  - 1 Exchange integration license
  - 1 MaxMobile license
  - 1 HMCP Combo Codecs
- The **Agent Combo license** bundle (ACM-COMBOAGENT-XX) includes the following licenses:
  - 1 Softswitch Station license
  - 1 IP Talk license
  - 1 Exchange integration license
  - 1 MaxMobile license
  - 1 ACM Agent seat
  - 1 MaxAgent seat
  - 1 MaxCall seat
  - 1 HMCP Combo Codecs
- The **Supervisor Combo license** bundle (ACM-COMBOSUPVR-XX) includes the following licenses:
  - 1 Softswitch Station license
  - 1 IP Talk license
  - 1 Exchange integration license
  - 1 MaxMobile license
  - 1 ACM Agent seat
  - 1 MaxAgent seat
  - 1 MaxCall seat
  - 1 MaxSupervisor seat
  - 1 HMCP Combo Codecs
  - 1 Agent Supervision
- Other Softswitch-eligible licenses:
  - SSW ACM Agent Seat license
  - MaxAgent Session and Seat licenses
  - MaxSupervisor Session and Seat licenses

- AltiConsole license
- CTI Integration Connector Seat license
- Call Recording Session and Seat licenses
- SIP Trunk license
- Third Party SIP Device license
- VRmanager license
- MaxInsight Session license
- Client Application SDK Session license
- Trunk Control APC SDK Session license
- SightMax Integration license
- Salesforce Connector license
- Polycom Advanced Features license

### Licenses Not Available

The following licenses are not available for the MAXCS All-In-One Softswitch Solution.

- IP Gateway Expansion licenses (ALTI-GWEXP-01).
- AltiStation licenses (you must use either a SSW-STATION stand-alone license or a SSW-BASECOMBO-STD license instead).
- ACM-SEAT500 (you must use either a SSW-ACMAGENT stand-alone license or a SSW-ACMCOMBOAGENT license instead).

## Minimum Requirements

- Dedicated Windows Server 2012 R2, Window 7 64-bit SP1, or 2008 R2 SP1 Standard running on a 64-bit, Quad-core Intel @2.5GHz, 4GB memory, 160GB hard drive, or, for a virtual environment, VMware ESX 5.5 or Hyper-V 6.1; allocate 4 Intel cores @ 2.0GHz each, 4GB memory, and 160GB hard drive
- The server must pass the tests in the HMCP Certification Tool; see Step 1: Run the HMCP Certification Tool on page 8.

**IMPORTANT:**

- Make sure the server has the recommended Windows Service Pack and/or updates.
- The server must already be a member of the Active Directory Domain.

### What Is HMCP?

HMCP stands for *Host Media Control Processing,* a virtual component that uses an Intel CPU to provide the following functions:

Process VoIP media stream

- Encode, decode, and transcode voice streams
- Detect and generate tone for IP devices
- Play music when a device is on hold
- Process IP paging

Play and record voice files

- Announce system and queue phrases
- Process auto attendant
- Process voice mail
- Call recording

Provide conferencing resources

- Barge-in/silent monitor/coaching
- Station conference and MeetMe conference

For a full explanation of HMCP and the roles it plays in the MAXCS All Software Solution, review the training presentation “What is HMCP?” in the AltiGen Communications Learning portal (http://learn.altigen.com/login/index.php). Log in with the same credentials that you use on the AltiGen Partner Portal.

You will run the AltiGen HMCP Certification tool before you begin the MAXCS installation. This tool helps measure how many channels can be simultaneously operated in certain conditions; you specify a threshold of CPU usage. It also measures DPC (Deferred Procedure Call) latency of Microsoft systems.

# Before You Begin

We recommend that you take these steps before you begin the installation process:

- Have the System key and software license keys available
- Make sure you have your login information for the Partner portal

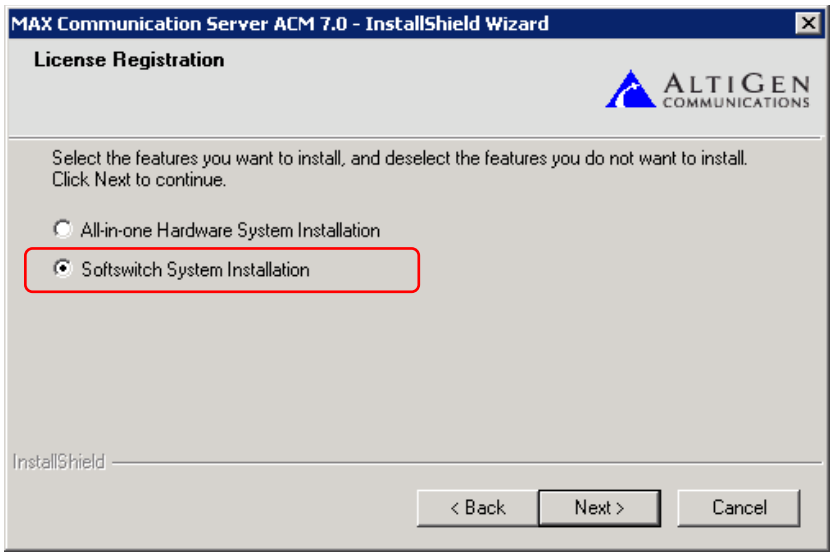
# Installation

This section provides detailed instructions for installing MAXCS ACM Softswitch Release 7.5.

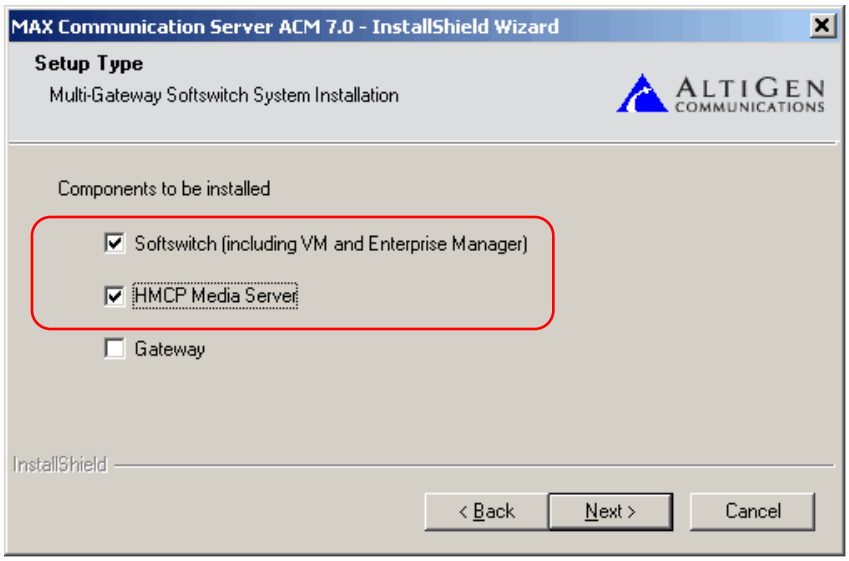
## Step 1: Install the HMCP Certification Tool

Before you install MAXCS, you will run the HMCP Certification Tool. This tool can be automatically installed during MAXCS installation.

1. Log into the server as a user with administrative privileges. If the system is a stand-alone server, log in as a local administrator account. If you plan to run Exchange Integration, you must have domain administrator rights.
2. Stop all MaxCS services.
3. Load the installation media. Open and review the readme.txt file.
4. Navigate to the *MAXCS ACM* folder. Double-click *Setup*. The installation wizard starts.
5. On the Welcome panel, close all Windows applications and click **Next**.
6. In the Setup Type panel, select **Softswitch System Installation**. Click **Next**.




7. On the next panel, select only the first two options, **Softswitch (including VM and Enterprise Manager)** and **HMCP Media Server**. Click **Next**.

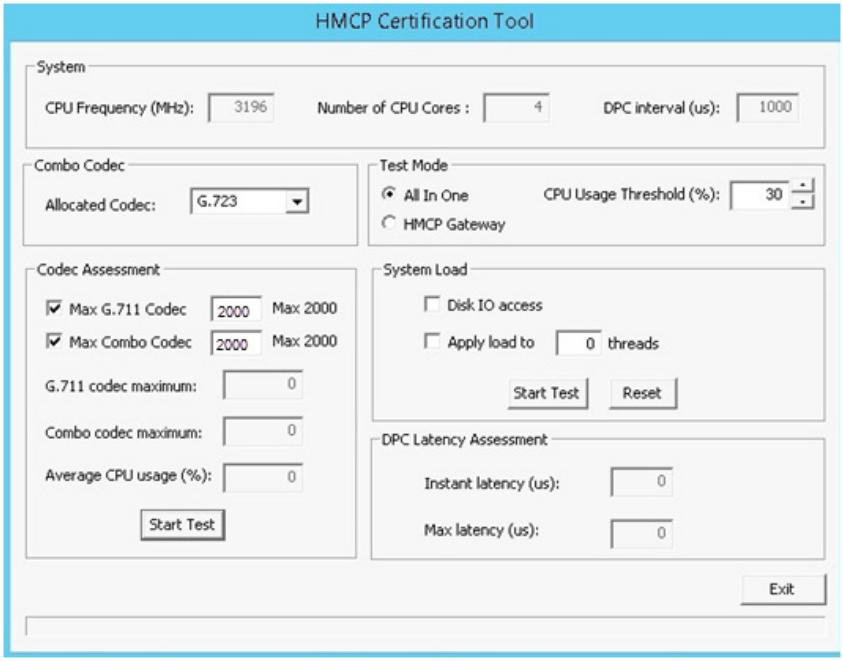



8. If you have not yet installed the MAXCS 7.5 HMCP Certification tool, you are prompted to do so now. You must run the certification tool before you install MAXCS. Click **Yes** to install the tool now.
9. The HMCP Certification tool installation wizard starts. Click **Next**.
10. In the next panel, enter a user name and organization, and then click **Next**.
11. In the next panel, click **Next** to accept the default folder for the installation.
12. In the next panel, click **Install**.

13. After the files have been installed, click **Finish**.
14. Click **Yes** to restart the server.

## Step 2: Run the HMCP Certification Tool

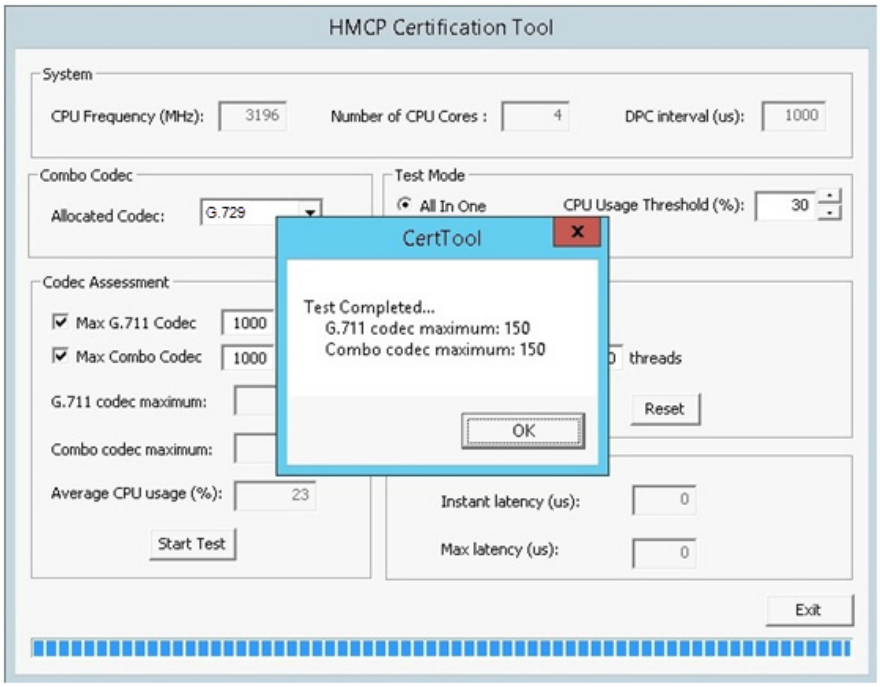
1. After the system restarts, click **Start** > **All Programs** > **AltiGen HMCP Certification Tool**.
2. Stop all AltiGen switching services.
3. When the HMCP tool opens, in the *Test Mode* section, select **All In One**. Set *CPU Usage Threshold* to **30**%.
4. In the *Codec Assessment* section, enter the maximum number of codecs in those two fields (enter 2000 for both Max G.711 and Max Combo)
5. Click **Start Test** in the *Codec Assessment* section.




6. Monitor the DPC Latency values. Instant latency should be less than 2000 us most of the time; Max latency should be less than 6000 us.
7. Record the number of supported G.711 and Combo codecs in the test results. Then enter those numbers in the *Max G.711 Codec* and *Max Combo Codec* fields (in the *Codec Assessment* section).
8. Under the *System Load* section, select both *Disk IO access* and *Apply load to xx threads*. Enter **1** for the number of threads. The tool will generate disk I/O access in the background and then measure DPC.
9. In the *Test Mode* section, set *CPU Usage Threshold* to **60**%.
10. Click the **Start Test** buttons in both the *Codec Assessment* section and the *System Load* section.
11. Monitor the DPC latency numbers. Instant latency should be less than 2000 us most of the time. Max latency should be less than 6000 us.

12. After the tool finishes its assessment, it shows the results. These values are the ***maximum*** quantities you should consider configuring. If you exceed these values, voice quality may start to degrade. Write down these values; you will refer to them later.

    Example: If you enter G.729 for the *Combo Codec* and set the *CPU Usage Threshold* to 30%, the tool will determine the optimal combination of G.711_only and Combo channels in the HMCP server.



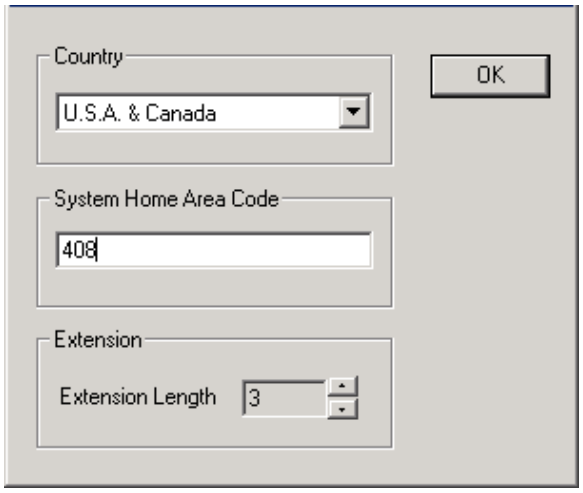

13. Click **OK** to close the results window. Click **Exit** to close the tool. Make sure that the tool has fully closed before you proceed; otherwise, the installation may not be successful.

## Step 3: Install MAXCS

Next, you will re-run the MAXCS installation program.

1. Navigate to the *MAXCS ACM* folder. Double-click *Setup*. The installation wizard starts.
2. On the Welcome panel, close all Windows applications and click **Next**.
3. In the Setup Type panel, select **Softswitch System Installation**. Click **Next**.
4. On the next panel, select only the first two options, **Softswitch** (including VM and Enterprise Manager) and **HMCP Media Server**. Click **Next**.
5. The next few panels prompt you to accept the License Agreement, enter your user name and company, specify a password, and choose a folder for the installation. Provide the requested information and click **Next** in each panel.
6. When you are prompted about the registration file, choose **Register Later**.
7. When prompted, accept the default locations for the CDR database and the PostOffice.
8. In the last panel, click **Install**.

9. If you are prompted to do so, install the software for KEYLOK, to support the soft system key: click **Install** and complete that installation process.
10. A warning indicates that you have not entered a system key. You will do this later; click **OK**.
11. When prompted, specify the country, the local area code for this server, and the number of digits you want for phone extensions. Click **OK.**

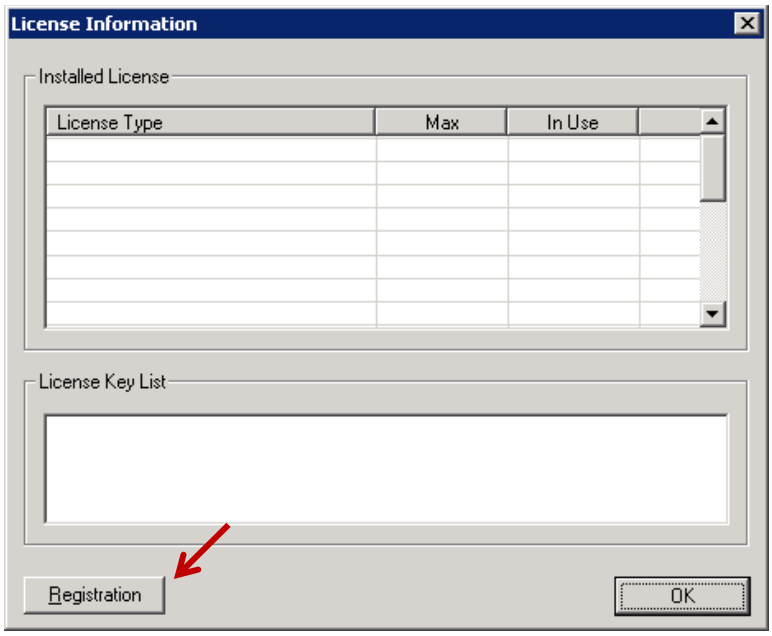



12. A window shows you the password; click **OK**.
13. You are reminded to upgrade other components, click **OK**.
14. When you are prompted whether to restart the system, select **Yes, I want to restart my computer now**, and then click **Finish**. The server reboots.

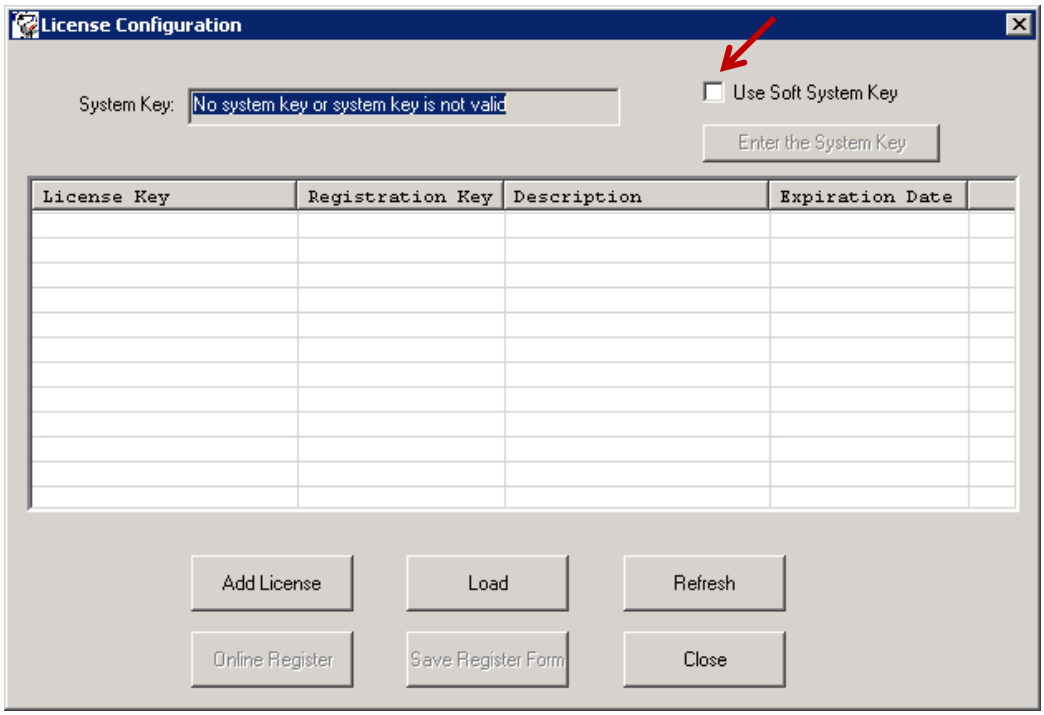
## Step 4: Add the System Key

Next, you will add the client's System Key to MaxAdministrator.

1. In Windows, click **Start** > **All Programs** > **MaxCommunication Server ACM** > **MaxAdministrator 7.5**
2. In MAXCS Administrator, choose the server.
3. When you are reminded that no system key has been provided, click **OK**.
4. Click the **Login** button and enter your password.
5. On the menu bar, choose **License** > **License Information**.
6. Click **Registration**.




7. In the License Configuration panel, check the box *Use Soft System Key*.
8. Click **Enter the System Key**.



9. Enter the System Key, which was provided by AltiGen Communications. This string usually begins with the prefix "SS." **You must enter this key in uppercase letters**. Click **OK**.
10. If you do not see the string in the field, click **Refresh**. The key will now appear in the System Key field.

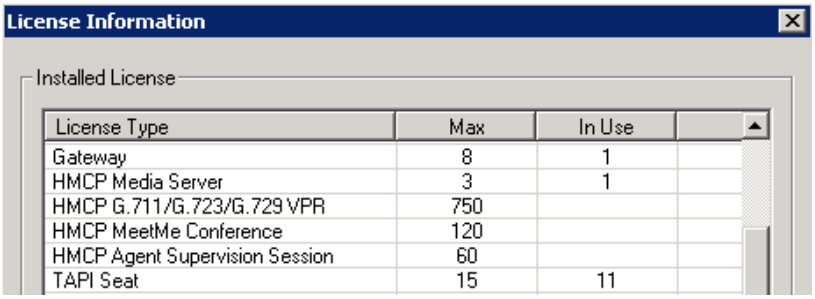
## Step 5: Register Licenses

Next, you will upload the license file(s) for this client.

1. In the License Configuration panel, click **Add License**.
2. Enter the **Softswitch Combo Base License** number and any other license numbers that you want to add.
3. If you have any Internet Explorer browsers windows open, close them at this point, and then click **Online Register**.
4. A new browser window opens. Click **Proceed to Online Registration**.
5. Log into the Partner site to view the *Assurance and Warranty* page.
6. On the first page, verify that the system configuration is correct. You can add more licenses here if you want to. Otherwise, click **Continue**.
7. On the next page, check **I have read and agree with the terms** to accept the **Assurance & warranty Agreement**, then click either **Continue With Price** or **Continue Without Price**.
8. Complete the contact information on the next page, and then click **Continue**.
9. View the *Assurance & Warranty Summary*, and then click **Continue Registration**.
10. On the final page, click **Download License Activation File** and save the EXCTL file somewhere on the server; you will load this file in step 13.
11. Close the web browser.
12. Return to the **License Configuration** window In MAXCS Administrator. Click **Load**.
13. Locate and load the EXCTL file that you downloaded in step 10.
14. The Softswitch Combo Base License should disappear, to be replaced by its component licenses.
15. Close the License Configuration window. Close the License Information window.
16. The licenses are now loaded, however the virtual boards do not appear because the license files were not in place when switching was started. To force these boards to appear, restart switching. (Use Windows **Start** > **All Programs** > **MaxCommunication Server ACM** > **Utilities** > **Start or Stop All AltiGen Services**.)

## Step 6: Configure the HMCP Virtual Board

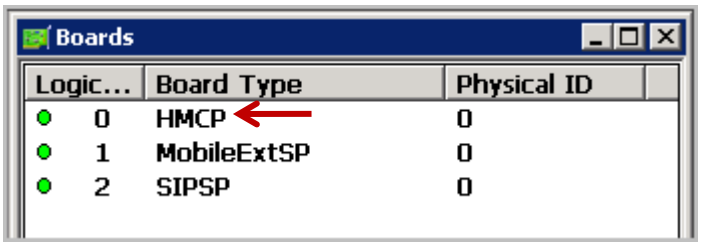
Next, you will configure the HMCP virtual board. An HMCP Media Server license is required to activate an HMCP virtual board.



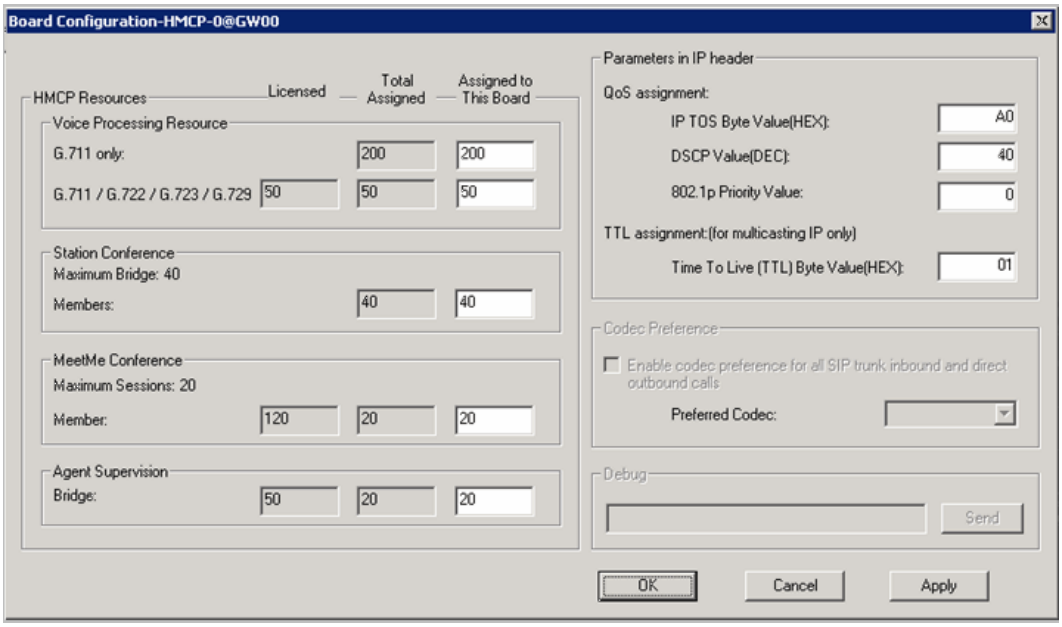

By default, the system grants 60 conference members in a maximum of 40 bridges.

You can change the number to as many as 120 members in a maximum of 40 bridges, and you can activate other HMCP resources.

1. In MAXCS Administrator, open the Boards panel and double-click **HMCP**.




2. Click **Board Configuration**.




3. Adjust the assigned number by entering a different number (up to the number your system is licensed for and not to exceed the maximum limit for each HMCP board) in the **Assigned to this board** fields and clicking **Apply**.

**Note:** Do not exceed the maximum numbers provided by the HMCP Certification Tool (from page 11).

The next section describes the parameters in this dialog box.

4. After adjusting these values, you may need to reboot the system so that the changes can take effect.

## HMCP Resources Parameters

This panel shows the total number licensed (if applicable), total currently assigned, and the number assigned to this HMCP board for Voice Processing Resources (VPR), station conference members, MeetMe conference members, and agent supervision bridges.

The maximum number of resources that can be supported on an HMCP virtual board is as follows:

- G.711 VPR - 1,000
- G.711 / G.722 / G.723 / G.729 VPR - 200
- Station Conference Members - 120
- MeetMe Conference Members - 120
- Agent Supervision Bridges - 20

**Notes**

**Important!** When adding additional combo licenses, the system will increase the RTP ports it uses and will use these new ports. If these additional ports are not added to the firewall, calls will not have audio. Please refer to the *MAXCS 7.5 ACM Administration Manual* Appendix “Network Ports” for instructions on how to identify these additional RTP ports.

- Codec G.722 is part of a combo codec and is controlled by license.
- 1,000 G.711 voice-processing resources will be licensed to the system when one HMCP Media Server license is registered.
- Do not assign more codecs than the system can support. Doing so can degrade system performance.
- The more VPRs you assign, the slower the system will be when it starts up. To calculate the optimized number of VPRs you need, use the following formula:
    - Total G.711 VPRs = Total number of local IP extensions
    - Total G.711/G.722/G.723/729 VPRs = Total number of remote IP phone users + Total Tie Trunk Channels that will use compressed codec

## Parameters in IP Header

This panel is where you configure QoS and TTL assignments.

- **QoS assignment** – IP TOS/DiffServ Byte Value. The default TOS/DiffServ byte hex value "A0" (10100000) signals the network switch and router that RTP packets are "Critical". To set the value for Diffserv Code Expedited Forwarding (DSCP EF), you can enter hex value "B8" (10111000).
- **TTL assignment** – For IP paging multicasting only. The purpose of the TTL (Time To Live) is to regulate how many hosts the IP paging packets can pass through. The TTL value is reduced by one on every hop. You may need to adjust this value if there are remote AltiGen IP phones at different locations that register to MAXCS through WAN and require the IP paging feature. The value will be the number of routers from MAXCS to remote IP phone plus one.

### *About Media Pass-Through Support*

One MaxCS 7.5 enhancement for system performance and voice quality is the *Media Pass-through* feature.

While the Media Pass-through feature is enabled, the HMCP driver does not need to do encoding and decoding on both channels.

In addition to benefits to direct calls, pass-through applies during call recording, silent monitoring, and coaching.

G.722 pass-through is enabled by default.

Media pass-through cannot support all conditions in the HMCP system, even if both connected channels are using same codec. The following MAXCS features do not supported media pass-through:

- Call playing – The trunk call playing must use encoder and decoder on RTP channel
- Conference calls – All RTP channels in a conference bridge must be encoded and decoded voice

## Step 7: Configure HMCP Codec Preferences for SIP Trunk Calls

**Note:** An on-premise Softswitch deployment may not need this step unless the machine is low power.

MaxCS 7.5 includes a feature designed to help you reduce the CPU consumption that occurs are a result of codec encoding and decoding. It allows you to indicate a codec preference (G.729 or G.711) for calls handled via a SIP trunk. This approach eliminates the steps of encoding and decoding - packets are directly forwarded to the endpoint.

The preference that you set must be supported by the SIP Trunk provider, and must be included in the codec profile list for SIP Trunks.

While this feature is enabled, the SIP device's codec configuration in *Enterprise Manager* will be ignored.

All SIP devices must support G.711 uLaw; if there are no common codecs on the device side, then G.711 uLaw will be used.

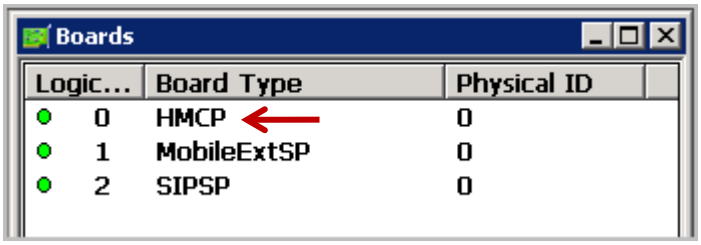
This codec preference applies to all SIP Trunk inbound calls and direct extension to SIP trunk outbound calls that are initiated from a phone or dialed via a MAXCS client.

Fax-over-IP overrides this setting; it will always use G.711.

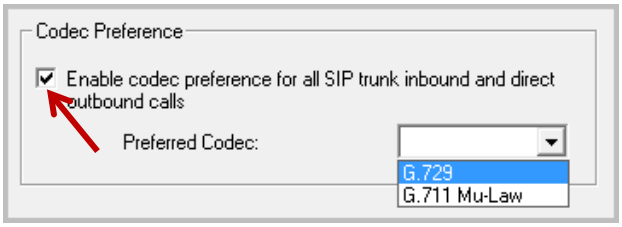
Outbound calls initiated by the system (such as calls from the voicemail system or ONA) may not use the preferred codec.

To configure this feature,

1. Log into MAXCS Administrator with the superpassword. If you do not log in via the superpassword, then the feature will not be enabled.
2. Open the Boards panel and double-click **HMCP**.



3. To enable the feature, select the checkbox and specify which codec to use (G.729 or G.711 Mu-Law). Click **Ok**.




## Codec Preference - Incoming Calls

When the Codec Preference feature is enabled, the system uses the following logic for handling incoming calls, based upon the content of the first SIP INVITE request.

| Incoming Calls - Preferred Codec G.729 | |
|---|---|
| **First SIP Invite Request Content** | **Codec Used for the Call** |
| SIP Trunk supports G.729 | The IP Codec table of Enterprise Manager is ignored. The system uses codec G.729 to negotiate with the endpoint. |
| SIP Trunk does not support G.729 | No codec is enforced. |

| Incoming Calls - Preferred Codec G.711 Mu-Law | |
|---|---|
| **First SIP Invite Request Content** | **Codec Used for the Call** |
| SIP Trunk supports G.711 Mu-Law | The IP Codec table of Enterprise Manager is ignored. The system uses codec G.711 Mu-Law to negotiate with the endpoint. |
| SIP Trunk does not support G.711 Mu-Law | No codec is enforced. |

If the target extension or DNIS number is enabled for Fax-over-IP (FoIP), then no codec is enforced.

## Codec Preference - Outgoing Calls, Third-Party IP Phones

When the Codec Preference feature is enabled, the system uses the following logic for handling outgoing SIP trunk calls from third-party IP phones. (Calls are considered SIP trunk calls if the target has a SIP trunk access code or an outcall routing access code prefix.)

| Outgoing Calls, 3rd-Party IP Phones - Preferred Codec G.729 | |
|---|---|
| **First SIP Invite Request Content** | **Codec Used for the Call** |
| Supports G.729 | The IP Codec table of Enterprise Manager is ignored. The system uses codec G.729 to negotiate with the endpoint. |

| | |
|---|---|
| Does not support G.729 | No codec is enforced. |

| Outgoing Calls, 3rd-Party IP Phones - Preferred Codec G.711 Mu-Law | |
|---|---|
| **First SIP Invite Request Content** | **Codec Used for the Call** |
| Supports G.711 | The IP Codec table of Enterprise Manager is ignored. The system uses codec G.711 to negotiate with the endpoint. |
| Does not support G.711 | No codec is enforced. |

If the source extension is enabled for Fax-over-IP (FoIP), then no codec is enforced.

### *Codec Preference - Outgoing Calls, IP Phones*

Because IP Phone SIP call requests are always from MAXCS, the behavior is different from calls from third-party phone extensions.

When a user makes an outgoing call from an IP phone, the system follows the IP Codec table in Enterprise Manager. All IP phones support G.729 and G.711 Mu-Law.

When the Codec Preference feature is enabled, the system uses the following logic for handling outgoing SIP trunk calls.

| Outgoing Calls, IP Phones | |
|---|---|
| **Preferred Codec** | **Codec Used for the Call** |
| G.729 | The system modifies the IP phone's codec in the RE-INVTE SDP body with G.729 codec. After the modification, MaxCS must pick G.729 and G.711 codecs as the preferred codecs to negotiate with the endpoint. |
| G.711 Mu-Law | The system modifies the IP phone's codec in the RE-INVTE SDP body with G.711 codec. After the modification, MaxCS must pick G.711 and G.729 codecs as the preferred codecs to negotiate with the endpoint. |

If the source extension is enabled for Fax-over-IP (FoIP), then no codec is enforced.

## Step 8: Configure the SIPSP Virtual Board

A VoIP connection typically consists of two parts.

**Signal Channel** – Sets up and tears down a call using protocol. SIP protocol is used in MAXCS to build a signal channel between the server and the IP phone.

**Media Path** – Encodes, transmits, and decodes voice for both parties.

The purpose of the virtual board SIPSP is to build signal channels for different connection types, IP extensions, SIP Tie Trunks, and SIP Trunking from ITSP. Each channel will have its channel ID similar to channels on a Triton extension or trunk board.

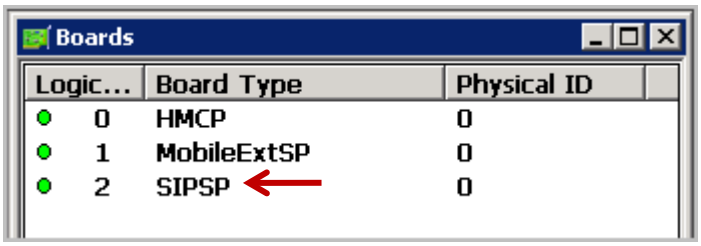
When an IP phone registers to the system, a channel ID will be assigned to the IP extension. However, these channels are only responsible for processing protocol and call control signals. They require a media path from a VoIP board or from the IP phone to establish a voice steam so that both sides can hear.

**Notes**

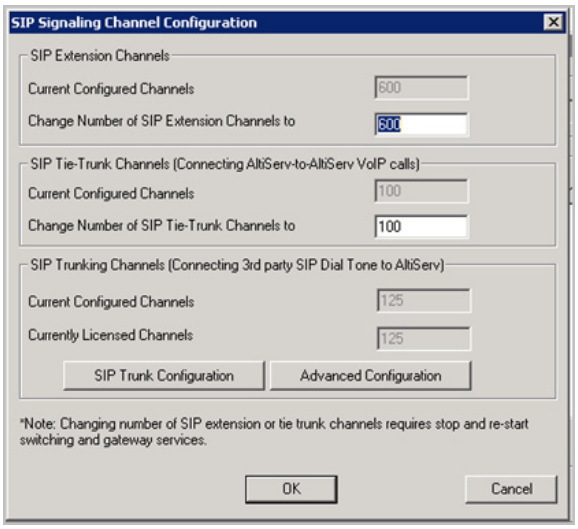
- Make sure you have enough VoIP resource boards.
- The more signal channels in the system, the more system memory and CPU power is required.
- Changing the number of signal channels requires that you restart the switching and gateway services.
- SIP Trunking Channel requires a license to activate.

To configure the SIPSP board,

1. In MAXCS Administrator, open the Boards panel and double-click **SIPSP**.




2. Click **Board Configuration.** The SIP Signaling Channel Configuration panel opens. This panel shows the number of configured channels and licensed channels.




3. Adjust the number of SIP extension channels and Tie-Trunk channels as needed.

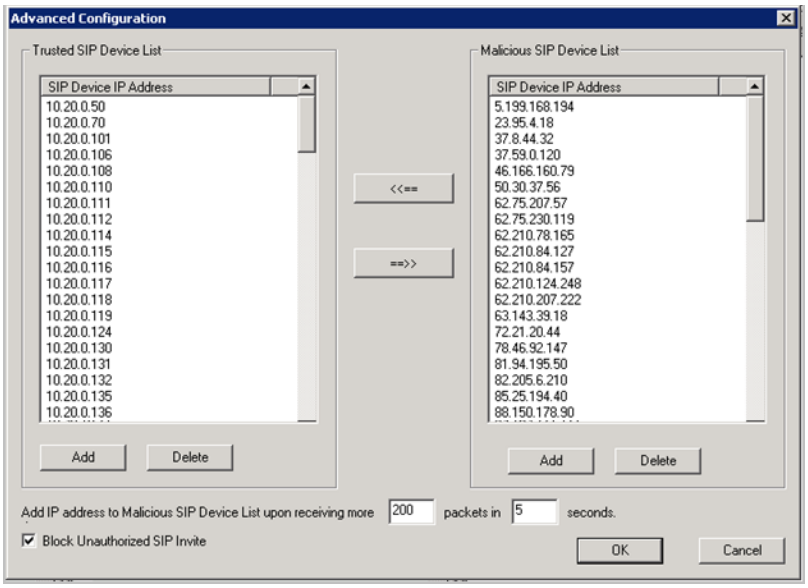
   By default, MAXCS is set to support 60 SIP extension channels. The maximum number possible depends of the system CPU performance, call volume, and usage.

   If a high performance machine is used as the Softswitch server, the number of channels can be more than 1000.

4. After you adjust these values, you must restart the switching and gateway services for this change to take effect. After the services restart, the new configuration will appear in the **Currently Configured Channels** fields.

5. After restarting the services, open MAXCS Administrator, double-click **SIPSP** in the Boards panel and then click **Advanced Configuration** to manage the *Trusted SIP Device* list.

   To move an IP address from one list to the other, select the IP address and click either the right or left arrow button.




## Step 9: Block Unauthorized SIP Invite Messages

You can perform this step now, or return and configure these settings later.

You can block unauthorized SIP invite messages; this setting is disabled by default.

When you enable the setting, SIP Invite requests are ignored if the IP address is not configured in one of the following places:

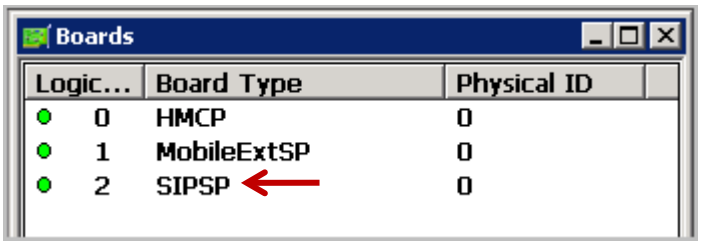
- SIP trunk
- IP dialing Table
- Trusted SIP Device List

IP extensions for the following devices are added to the Trusted SIP Device list automatically, once they successfully register to the system (unless they are found in the Malicious SIP Device list:

- AltiGen IP Phones
- Third-Party SIP Devices
- IPTalk

To enable protection,

1. In MAXCS Administrator, open Boards view and double-click **SIPSP**.




2. Click **Board Configuration**, and then click **Advanced Configuration**.
3. Check *Block Unauthorized SIP Invite* and click **OK**.

## Step 10: Configure the SIP Trunks

Next, you need to configure SIP trunk channels. Before you start, note the following:

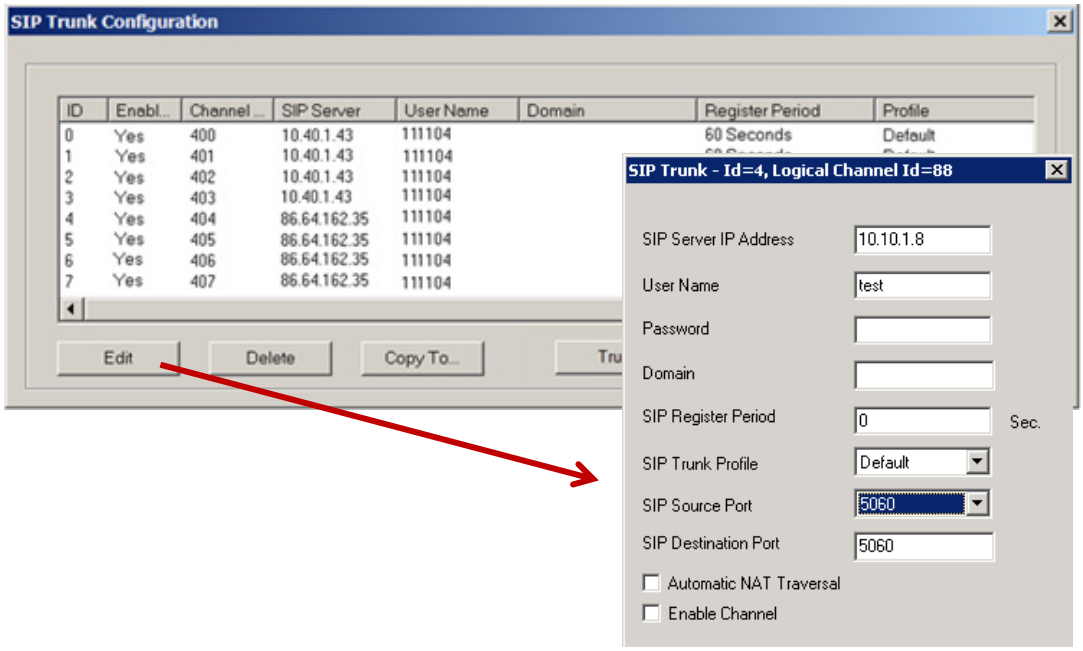
- An AltiGen SIP Trunking channel is licensed. You need to buy and register a license to be able to configure this option.
- AltiGen does not guarantee the voice quality of the SIP dial tone coming from your service provider. You need to work with your data service and SIP trunking service provider to make sure adequate QoS is provisioned for your WAN service.
- AltiGen does not guarantee SIP trunk implementation will work with all SIP dial tone service providers. AltiGen dealers are notified of AltiGen-tested and certified SIP-Trunk service providers. Configuration guidelines for each AltiGen-certified SIP-Trunk service provider can be found in the AltiGen authorized Partner Knowledge Base, available from the AltiGen Partner Web Site. SIP dial tone service providers need to support the following:
    - G.711, G.723.1, G.729 codec
    - RFC 2833 for DTMF tone delivery
    - SIP MD5 authentication with SIP registration
- If MAXCS is behind NAT, verify that your SIP SP can support this configuration.

When subscribing to a SIP dial tone service, typically your service provider will provide you with the information required in the SIP Trunk Configuration dialog box. Enter the service parameters for each SIP trunk channel configuration individually.

**Note:** Make sure you have enough IP resource boards to cover your needs.

To configure a SIP trunk,

1. In MAXCS Administrator, open the Boards panel and double-click **SIPSP**.
2. Select a SIP trunk type and click the **Trunk Properties** button.
3. Click **the SIP Trunk Configuration** button.




4. Edit or fill in parameters as needed. To edit a line, click **Edit** and update the entry). Click **OK**.
   - SIP Server IP Address – The SIP Trunk service provider's server IP address.
   - User Name – Assigned by the SIP Trunk service provider.
   - Password – Assigned by the SIP Trunk service provider.
   - Domain – The Domain Name of the SIP Trunk service provider, if required.
   - SIP Register Period – How frequently the AltiGen system needs to send SIP registration packets to the service provider. This can detect if the service provider is up or not. Some service providers do not accept SIP Register messages. In these cases, you can disable sending SIP Register messages from MAXCS by setting the SIP Register Period to 0.
   - SIP Trunk Profile – Select the appropriate SIP trunk profile.
   - SIP Source Port – For SIP UDP, select the source port from 5060 or 10060. For TCP or TLS, you cannot change ports. Using a port other than 5060 will prevent SIP-ALG firewall/router from changing the SIP packets. Port 10060 is recommended, if the service provider supports it.
   - SIP Destination Port – A SIP Trunk can have different source port and destination port.
   - Automatic NAT Traversal – Leave this box unchecked.
   - Enable Channel – After all above parameters are entered correctly, check this box to activate the channel. The AltiGen system will send authentication to the service provider to verify the setting.
5. To copy the information in one row to other rows, select the row and click **Copy To**. Then select the rows you want to copy the information to, using **CTRL**+click and **Shift**+click to select several rows. Click **OK**.

   If you need to delete a row, select it and click **Delete**.

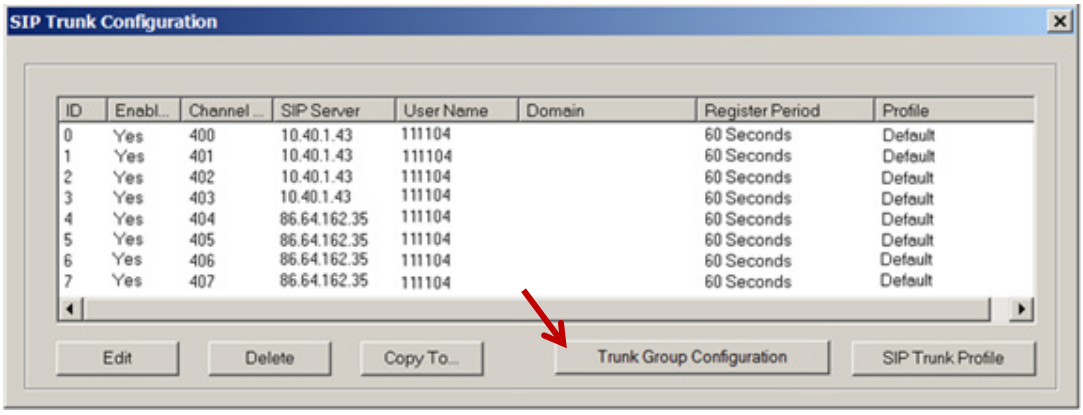
6. In Enterprise Manager, add the SIP Trunk service provider's IP address to the IP Device Range (on the **Servers** > **IP codec** tab) and select the proper codec profile for this service. Failure to do this step may result in no audio, even if the SIP Trunk channel shows the call is connected.

## Step 11: Configure SIP Trunk Groups

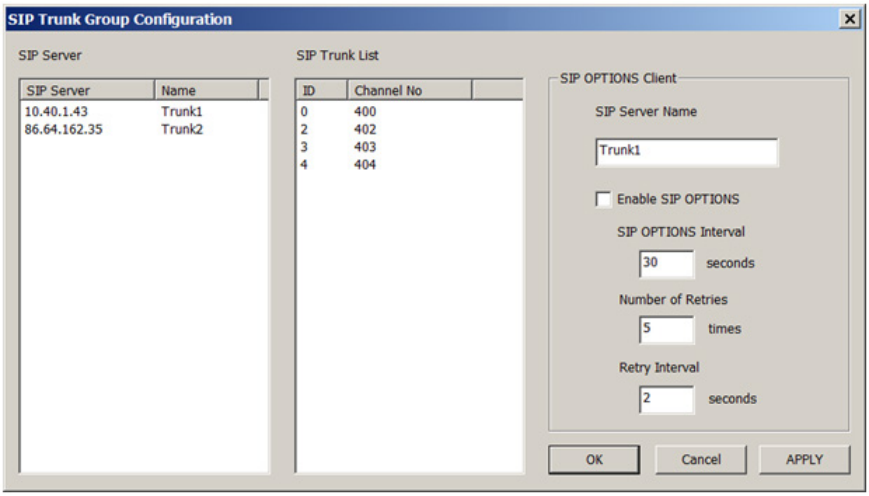
You can configure trunk groups so that SIP devices can exchange their status. By sending a "keepalive" message and checking for a valid response, SIP devices will know whether remote peers are ready to receive a new request.

Once the MaxCS server has been set up, SIP trunks will try to register to the SIP servers first. If the registration is successful and if the SIP Options feature is enabled, MaxCS will send the "keepalive" message to SIP providers according the interval setting. Only one "keepalive" message will be sent for each group of trunks.

1. In the *SIP Trunk Configuration* dialog box, click **Trunk Group Configuration**.




2. In the *SIP Trunk Group Configuration* window, select a SIP Server on the left, and then configure SIP Options for that SIP Trunk group as appropriate.




The window includes SIP provider IP address, SIP trunk List, and SIP Options settings. SIP trunks are grouped by SIP provider IP address; SIP trunks with the same SIP Server IP address are treated as a group.

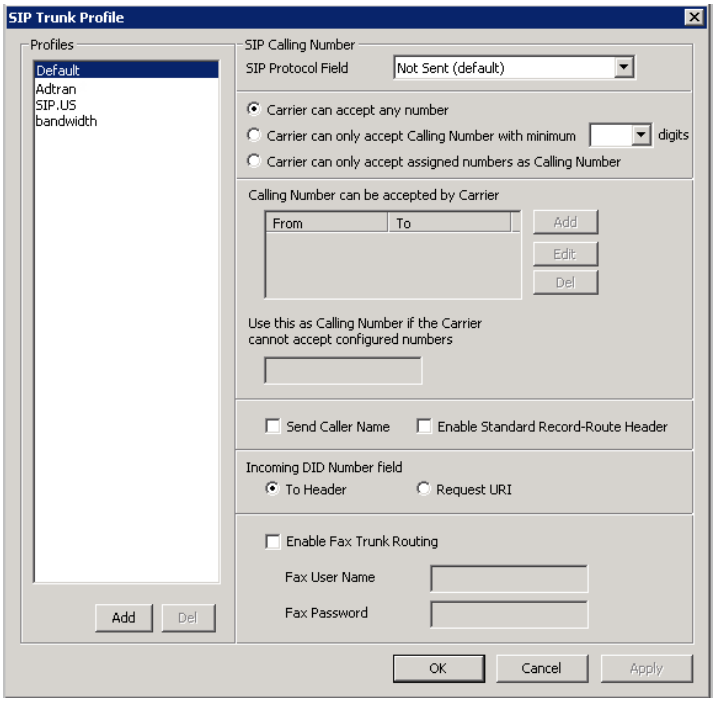
*SIP Options* can be enabled or disabled for each SIP Server. By default, *SIP Options* is disabled.

- SIP Server Name – A name for this SIP Server.
- Enable SIP OPTIONS – Enable or disable this feature for the selected SIP Trunk group.
- SIP Options Interval – How often, in seconds, the server sends a "keepalive" message to this enabled SIP trunk group. The default interval is 30 seconds.
- Number of Retries – If MaxCS receives no 200 (OK) response, the number of times a "keepalive' message should be sent. After these retries, if there still has been no valid response, then MaxCS marks all SIP trunks in the group as Not Ready. The default number of attempts is 5.
- Retry Interval – While a SIP trunk group is in a Retry state and is not receiving a valid response, how often MaxCS should send another "keepalive" message to the SIP server. The default interval is 2 seconds.
- If MaxCS does not receive a SIP 200 (OK) message after the set number of retries, it then sets all SIP trunks in that group to Not Ready.

## Step 12: Create a SIP Trunk Profile

Different SIP service providers may support different ways of sending a caller ID. To provide callees with a more accurate caller ID, you can create a SIP Trunk Profile for a particular service provider, when necessary. Otherwise, a default profile is used. Once you have created a profile, you can select it in the SIP Trunk Configuration panel.

1. In the *SIP Trunk Configuration* dialog box, click the **SIP Trunk Profile** button on the right.




2. Edit the fields as necessary and click **OK**. (Refer to the following table for descriptions of these fields).

| SIP Trunk Profile Field | Description |
|---|---|
| SIP Protocol Field | **Not Sent** (default) – Do not send transmitted caller ID<br>**FROM Header** – Send the caller ID using the SIP FROM header<br>**P-Preferred Identity** – Send the caller ID using the SIP P-Preferred Identity header<br>**P-Asserted Identity** – Send the caller ID using the SIP P-Asserted Identity header |
| Carrier can accept any number | This is the default. |
| Carrier can only accept Calling Number with minimum x digits | Enter the number of digits, and then enter a calling number in the field below the table in case the carrier cannot accept configured numbers. |
| Carrier can only accept assigned numbers as Calling Number | If you select this option, specify "assigned numbers" by clicking the **Add** button and entering the numbers. To edit or delete a number you added, select it and click the **Edit** or **Del** button. Enter a calling number in the field below the table in case the carrier cannot accept configured numbers. |
| Send Caller Name | Check to also send the caller name to callees. |
| Enable Standard Record-Route Header | Check this box if the SIP service provider uses SIP Record-Route and the SIP trunk cannot make or receive calls. If it already works, DO NOT CHECK or UNCHECK this box. [Service provider Bandwidth.com with Edgewater Route require this checked] |
| Incoming DID Number Field | When a call comes in, the SIP trunk uses **To Header** or **Request URI** as the DID/DNIS number |
| Enable Fax Trunk Routing | This feature is supported on AltiGen SIP trunks only.<br>If the extension is a fax extension and *Fax Trunk routing* is checked, that means the SIPSP should use the Fax Username and Fax password of SIP Trunk profile instead of regular username and password of SIP Trunk to negotiate with the SIP Trunk.<br>If the extension is a fax extension but *Fax Trunk routing* is unchecked, that means the SIPSP should use the regular username and password of the SIP Trunk to negotiate with the SIP Trunk side.<br>Fax User Name – The user name for fax routing<br>Fax Password – The password for fax routing |

## Step 13: Configure a Codec Profile

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

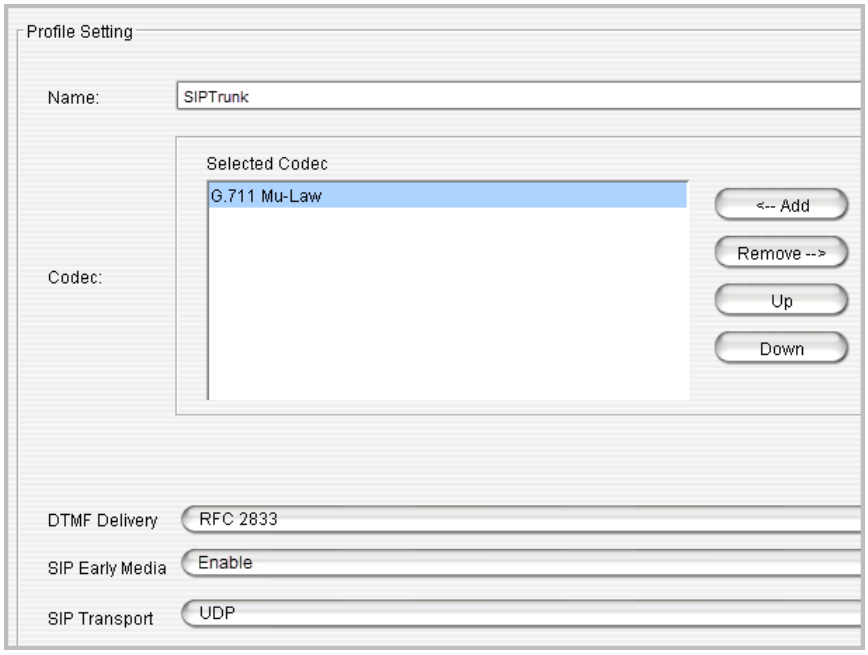
1. In MAXCS Administrator, select **VoIP** > **Enterprise Network Manager**.
2. Click the **Codec** button on the *Quick Launch* bar (this is different from the *IP Codec* tab).

3. Check whether a profile named *SIPTrunk* has been preconfigured for you.

   If it has already been configured, confirm that the settings are correct.

   If this profile was not preconfigured, click **Add** and create a profile with the following parameters:

   - For the name, enter **SIPTrunk**.
   - Set the *Selected Codec* to **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**
   - Click **Advanced** and set both packet lengths to **20** ms

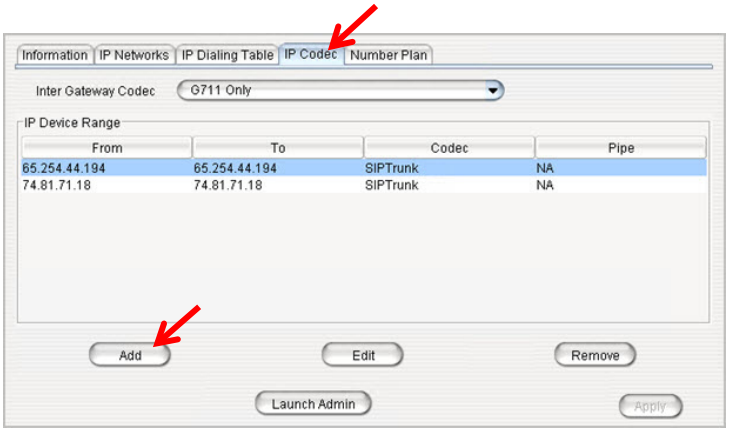


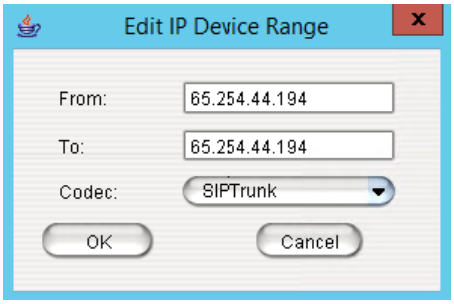

## Step 14: Assign the Codec Profile to the Two SIP Servers

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. Determine whether the following servers were preconfigured for you:
   - 65.254.44.194
   - 74.81.71.18
4. If these servers were preconfigured for you, confirm that each server is assigned to the Codec Profile *SIPTrunk*.

   If they were not preconfigured for you, click **Add**, add the first server, and set the Codec Profile to *SIPTrunk*. Repeat for the second server.

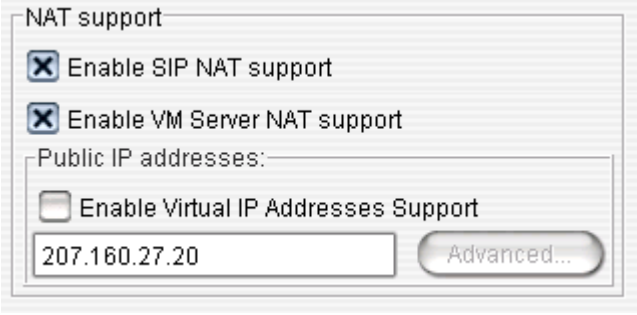



## Step 15: Configure NAT

To confirm that NAT support has been configured,

1. Click the **IP Networks** tab and confirm that both of the NAT support checkboxes are selected. If they have not been selected, check both boxes.

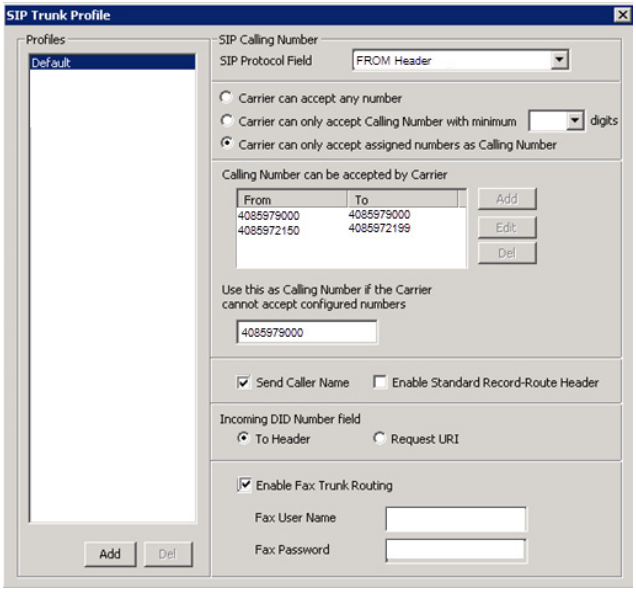



2. Close Enterprise Manager.

## Step 16: Add a New SIP Trunk Profile

We recommend that you add 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**.




2. Create a new SIP Trunk profile, specifying the following parameters.
    - Set the *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 **To Header**.

## Step 17: Channel Configuration

**Note:** These steps apply **only** to deployments using AltiGen SIP Trunks. Before you begin, retrieve the configuration details from the MaxCS Cloud Services portal, in the *SIP Services* section of the *General* tab of your account.

If your deployment uses SIP services from another AltiGen-certified SIP Trunk carrier, you will need to get the specific configuration details from that provider. Configuration details will vary from one provider to another.

For load balancing, 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 the following figure 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 | ← - - - ChannelB |
| 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: _____

1st Server channels: _____ - _____

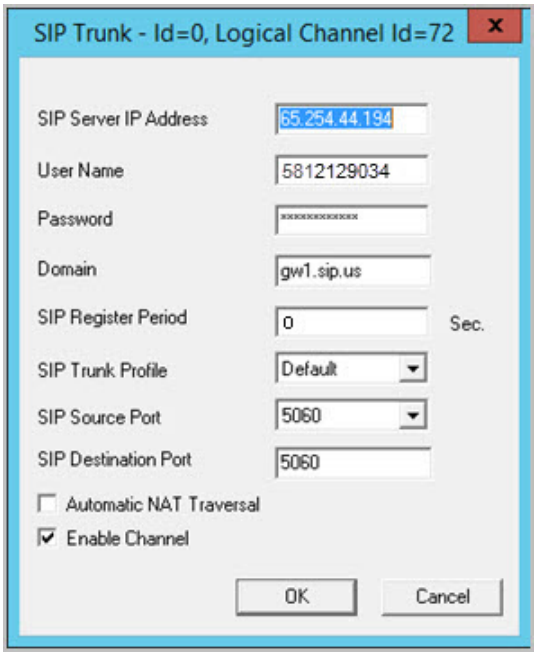
2nd Server channels: _____ - _____

### *Configure the First Half of the Channels*

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

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

2. Switch back to MAXCS ACM Administrator. You should be in the *SIP Trunk Configuration* page. If not, open it now.
3. Select the first SIP Trunk channel and click **Edit**. Set the following parameters and click **OK**:
   - For *SIP Server IP Address*, enter the SIP Trunk IP address of the first SIP trunk server: 65.254.44.194.
   - For *User Name*, enter the *SIP Trunk Username* number (from the order in the Cloud Services portal). In our example, this username is *5812129034*.
   - For *Password*, enter the *SIP Trunk Password* (from the order in the Cloud Services portal). Click **Hide/Show Password** to see it.
   - For *Domain*, paste the *URL* (from the order in the Cloud Services portal): *gw1.sip.us*.
   - 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**.




4. 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 our example, you would copy the configuration of channel 00 to channels 01 through 09. Click **OK**.
5. 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**.

## *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 the previous step, except that you are using the "Secondary" SIP details (from the order in the Cloud Services 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, 74.81.71.18. This is different from the IP address for the first set of channels.
   - For *User Name*, paste the same string as for the first set of channels. In our 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 our example, this is *gw2.sip.us*.
- 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**.

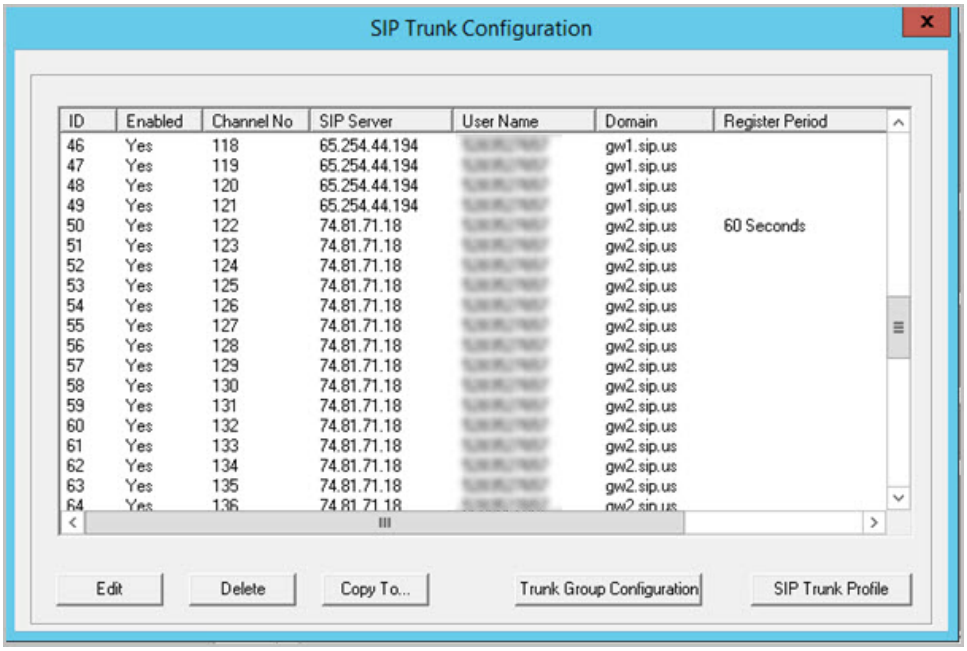
3. Copy the configuration of ChannelB to the rest of the ***last half*** of the channels:
   a. Select ChannelB and click **Copy To.**
   b. Select the range of channels from the next channel (after ChannelB) through all of the rest of the channels. In our example, you would copy the configuration of channel 10 to channels 11 through 19. 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**.

## *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 the next figure, 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.

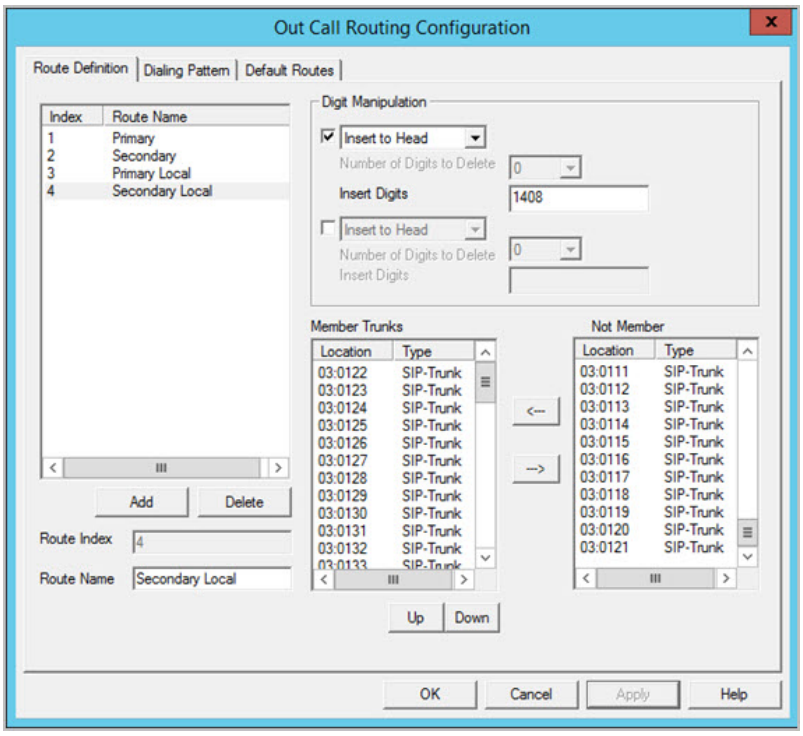



3. Return to the main Trunk view (close the various panels to reach that view) and confirm that all configured SIP trunk channels show as *Idle*, including the two channels that are configured

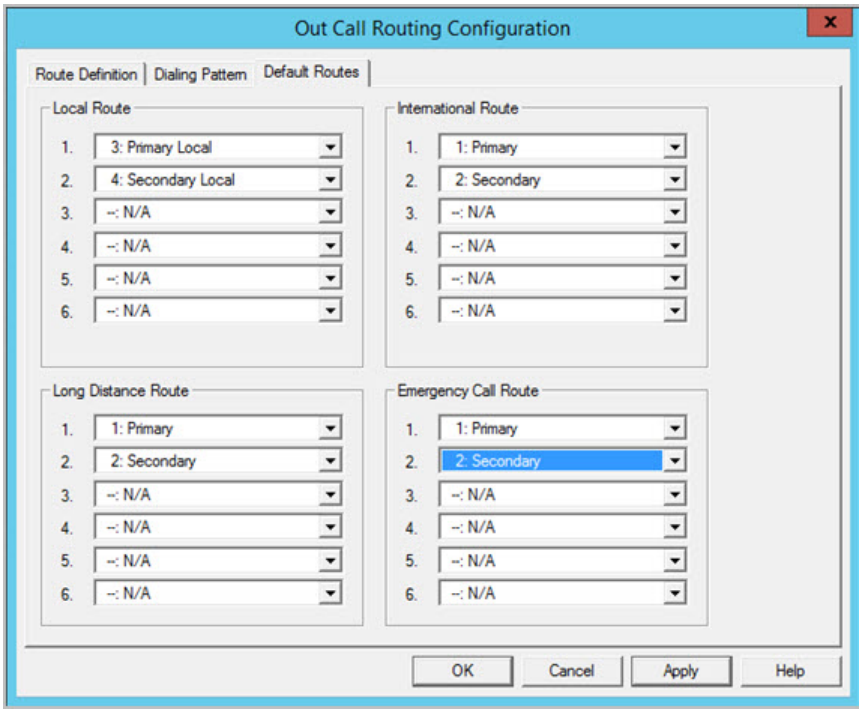
## Step 18: Out Call Routing

You can configure out-call routing rules; this step is optional. Refer to the *Out-Call Routing Configuration* chapter of the *MAXCS 7.5 ACM Administration Manual* for full details.

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 30, 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**.




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



- 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**

## Step 19: Configure Inbound Routing

Carriers send 11 digits as DNIS; configure your inbound routing rules accordingly.

## Step 20: System Configuration

At this point, calls should come into the system. Your next steps are to configure out-call routing and other internal processing rules.

Refer to the *MAXCS 7.5 ACM Administration Manual* for instructions:

- Chapter 4, System Configuration
- Chapter 26, Enterprise Manager

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

## Step 21: Configure Polycom Phones

If the organization has Polycom phones, refer to the *MAXCS 7.5 Polycom Phone Configuration Guide* for instructions on how to configure those IP phones. This document is available in the AltiGen Knowledge Base.

**Note:** Configuring Polycom phones in Release 7.5 is very different from the configuration process used in previous releases.

# AltiGen Technical Support

AltiGen provides technical support to Authorized AltiGen Partners and distributors only. End user customers, please contact your Authorized AltiGen Partner for technical support.

**Authorized AltiGen Partners** and distributors may contact AltiGen technical support by the following methods:

- You may request technical support on AltiGen's Partner web site, at https://partner.altigen.com. Open a case on this site; a Technical Support representative will respond within one business day.
- Call 888-ALTIGEN, option 5, or 408-597-9000, option 5, and follow the prompts. Your call will be answered by one of AltiGen's Technical Support Representatives or routed to the Technical Support Message Center if no one is available to answer your call.

Technical support hours are 5:00 a.m. to 5:00 p.m., PT, Monday through Friday, except holidays.

If all representatives are busy, your call will be returned in the order it was received, within four hours under normal circumstances. Outside AltiGen business hours, only urgent calls will be returned on the same day (within one hour). Non-urgent calls will be returned on the next business day.

Please be ready to supply the following information:

- Partner ID
- AltiGen Certified Engineer ID
- Product serial number
- AltiWare or MAXCS version number
- Number and types of boards in the system
- Server model
- The telephone number where you can be reached