MAXCS Release 7.5

MultiVoIP Multitech Configuration Guide

Intended audience:

AltiGen Authorized Partners

July 7, 2015

Contents

Introduction

This document provides guidelines for setting up a MultiTech gateway as a SIP trunking gateway, and for configuring SIP extensions. This document uses MVP410 as an example. The same configuration should also apply to other models of the MultiTech MultiVoIP gateway.

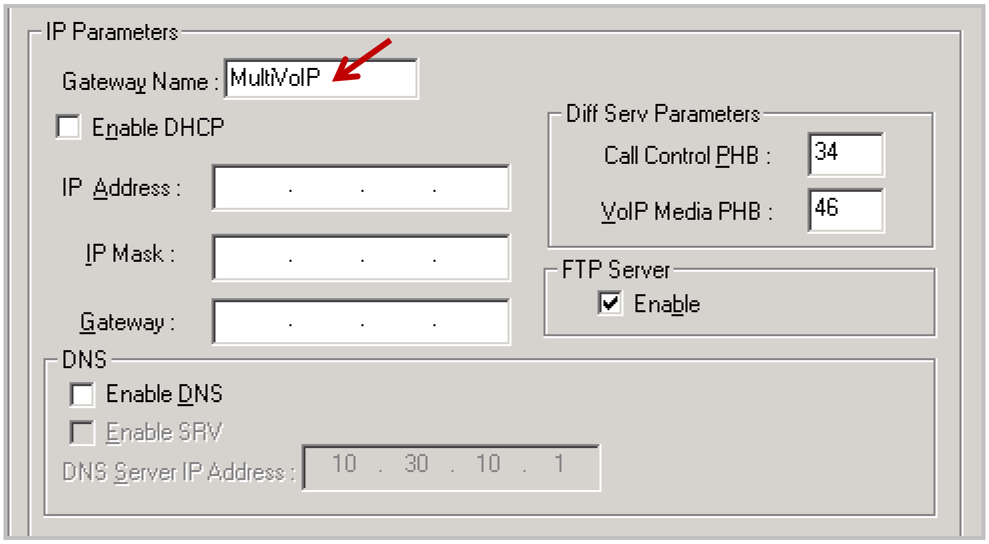
The firmware version for MVP410 used in this example is 6.09.0a. The AltiGen IP phone firmware version needs to be 2xxx (SIP-based).

Requirements

- Your system must be running MaxCS release 7.5.
- You must have a MultiTech MultiVoIP Voice over IP gateway (MVP210, MVP410, or MVP810)
- Both the MaxCS system and the MultiVoIP gateway need to have public IP addresses or be in the same private network range. NAT is not supported at this time.
- If private network addresses are used, an intranet (such as VPN, frame-relay, and so on) is required prior to configuration of the remote MultiVoIP gateway.
- The MultiVoIP gateway's IP address is configured as 10.10.101.81 in this example. Please consult the MultiVoIP documentation about gateway IP address configuration.

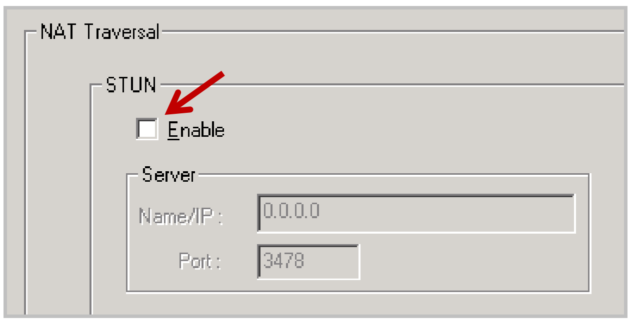
MultiVoIP Gateway Configuration

1. Log into the MultiVoIP configuration tool. Refer to your product documentation for the URL and any required username or password.
2. After the IP address is configured, in the MultiVoIP configuration tool select **Configuration** > **Ethernet/IP** and assign a **Gateway Name**. In our example we use the name **MultiVoIP**. (Do not leave this field empty.)



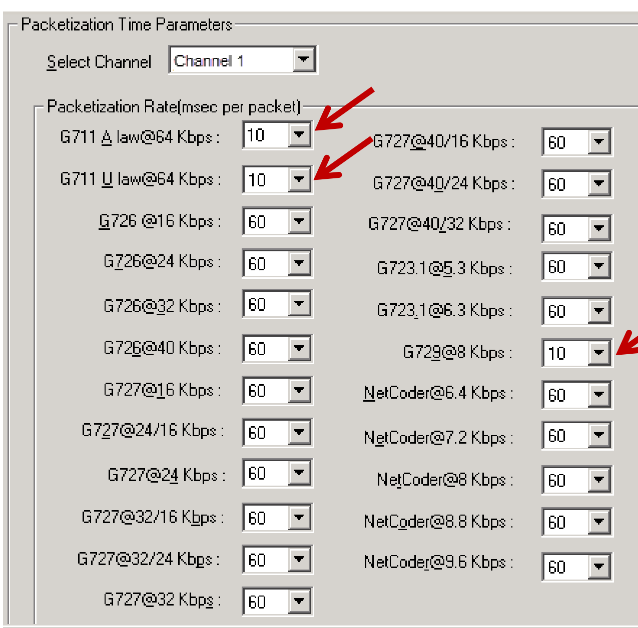
3. Select **Configuration** > **Call Signaling** > **SIP**. Set *Signaling Port* to **5060**.

4. Select **Configuration** > **Call Signaling** > **NAT Traversal**. In our example, we do not use the STUN protocol, so the checkbox is cleared.

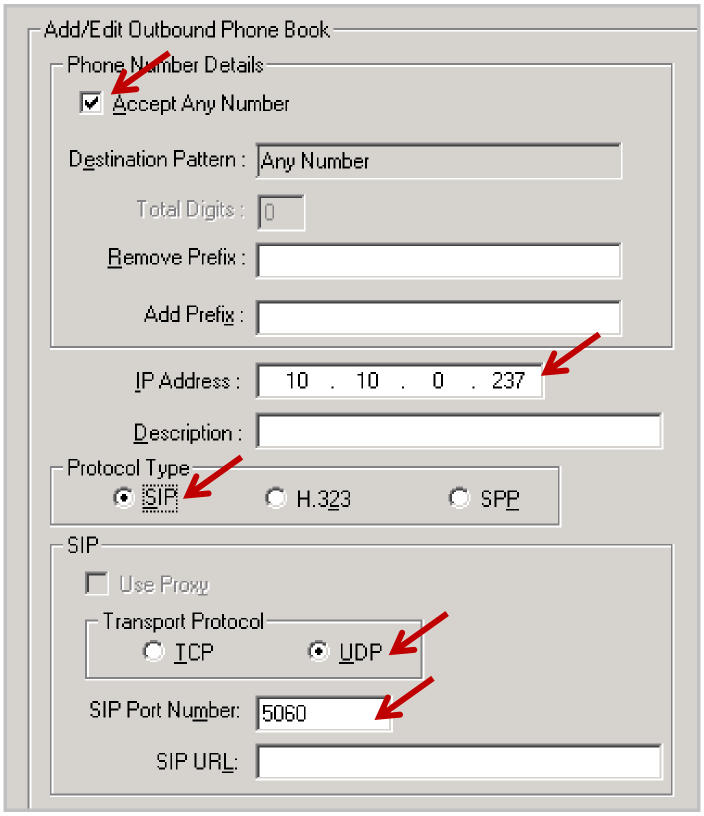


5. Select **Advanced** > **Packetization Time**.
   - Set *G.711 A-law@64Kbps* to **10** (ms)
   - Set *G.711 U-law@64Kbps* to **10** (ms)
   - Set *G.729@8Kbps* to **10** (ms)

   Important – if these parameters are not set to 10ms, you may experience one-way-only audio.
6. Click **Copy Channel** to copy the setting to all the channels.

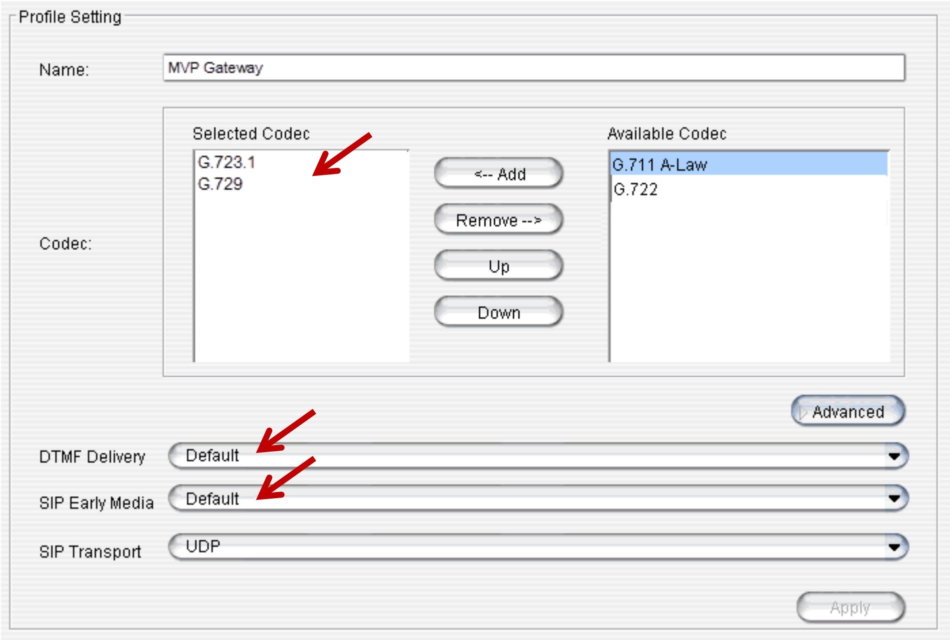


7. Select **Phone Book** > **Outbound Phone Book**.
8. Either create a new entry or edit an existing entry with the following parameters:
   - Check *Accept Any Number*.
   - Set *IP Address* to the MaxCS IP Address, **10.10.0.237**.
   - Set *Protocol Type* as **SIP**.
   - Set *Transport Protocol* as **UDP**.
   - Set *SIP Port Number* as **5060**.

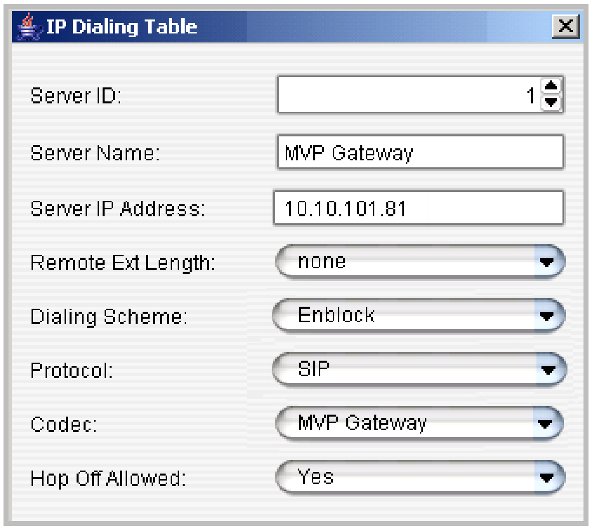


Enterprise Manager Configuration

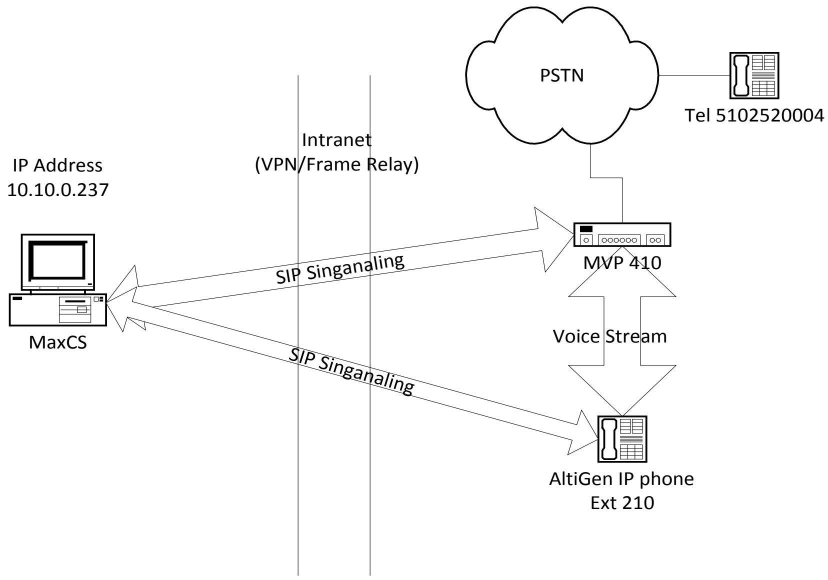
1. Open **Enterprise Manager** and log in.
2. Select the **Codec** tab and the **IP Codec** sub-tab.
3. Add a new codec profile named **MVP Gateway**. Set the following parameters:
   - Use the **Add** and **Remove** buttons to move *G.723.1* to the first entry in the *Selected Codec* list, followed by *G.729*.)
   - Set both *DTMF Delivery* and *SIP Early Media* to **Default**



4. Select **Enterprise Manager** > **IP Dialing Table**.
5. Add an entry (**Server ID** is 1 in this example) with the following parameters:
    - Set *Server Name* to **MVP Gateway**
    - Set *Sever IP Address* to **10.10.101.81**
    - Set *Dialing Scheme* to **Enblock**
    - Set *Protocol* to **SIP**
    - Set *Codec* to **MVP Gateway**



The “Connect Voice Stream to Server” Option



In the figure above, a call comes from PSTN to the gateway and connects to AltiGen IP phone 210.

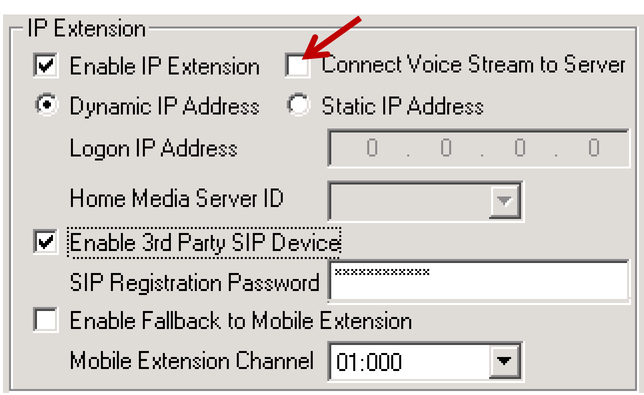
There are two SIP signaling paths. The first signaling path is between the MultiVoIP gateway and MaxCS. The second path is between the AltiGen IP phone and MaxCS.

The voice stream goes directly between the MultiVoIP gateway and the AltiGen IP phone, so that the voice quality is better and the latency is low. This path barely consumes intranet bandwidth. This is the default configuration.

Here is an alternative configuration:

In MaxCS Administrator, select **PBX** > **Extension Configuration**.

If the **Connect Voice Stream to Server** option is checked, then the voice steam will always go back to the server. In this configuration, the path will consume more intranet bandwidth, the voice quality will be become degraded, and the latency will become higher.



Under the following conditions, the system administrator will need to check the **Connect Voice Stream to Server** option:

- When the Voice Recording feature is used on the AltiGen IP phone (or the FXS extension behind MVP410)
- When a supervisor wants to silently monitor, barge in to, or coach the extension

In these cases, the administrator needs to make sure the bandwidth between headquarters and the branch office is sufficient.

Basic Configuration

The MVP410 has 4 FXS/FXO channels. Each channel can be configured as a SIP extension or SIP tie-trunk channel to PSTN. It is possible to configure two channels as two SIP extensions and two channels as two SIP trunking channels to PSTN.

Important! **FXS channels must be configured first.** For example, if your device has four channels and you want to provision two as analog extensions and the other two channels as SIP trunking channels, then you must configure channels 1 and 2 as FXS ports, leaving channels 3 and 4 for the trunks. If you do not provision the first two channels as the analog channels, then you may encounter one-way-only audio issues**.**

SIP Extension Configuration

Before you configure SIP extensions, you must check that all of the requirements in the Prerequisites section have been met.

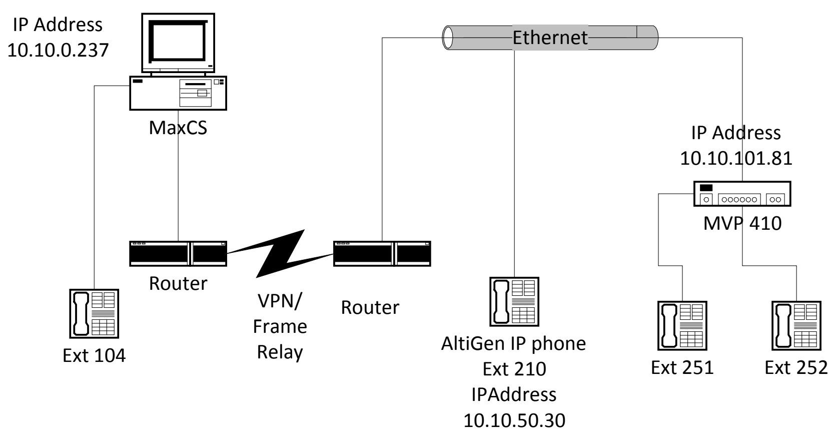


Figure 1: SIP Extension Configuration

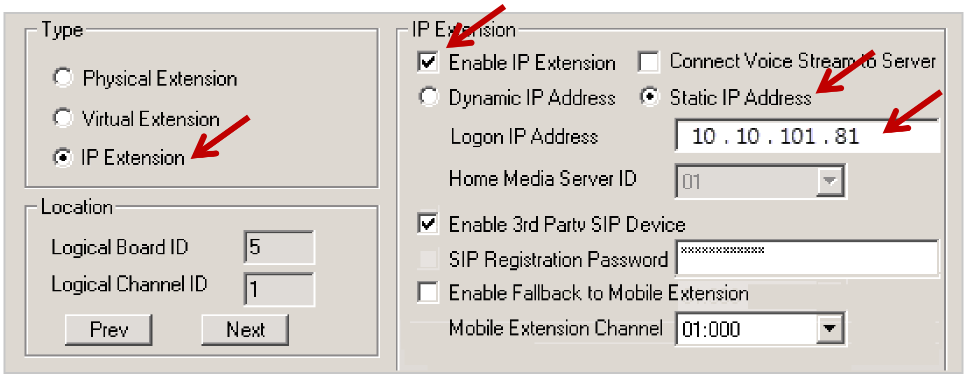
This section shows steps for the following example configuration:

- The IP address of MaxCS at headquarters is 10.10.0.237
- MaxCS has one analog extension, extension 104
- In the remote site, the IP address of the AltiGen IP phone at extension 210 is 10.10.50.30
- The IP address of the MultiVoIP MVP 410 gateway is 10.10.101.81

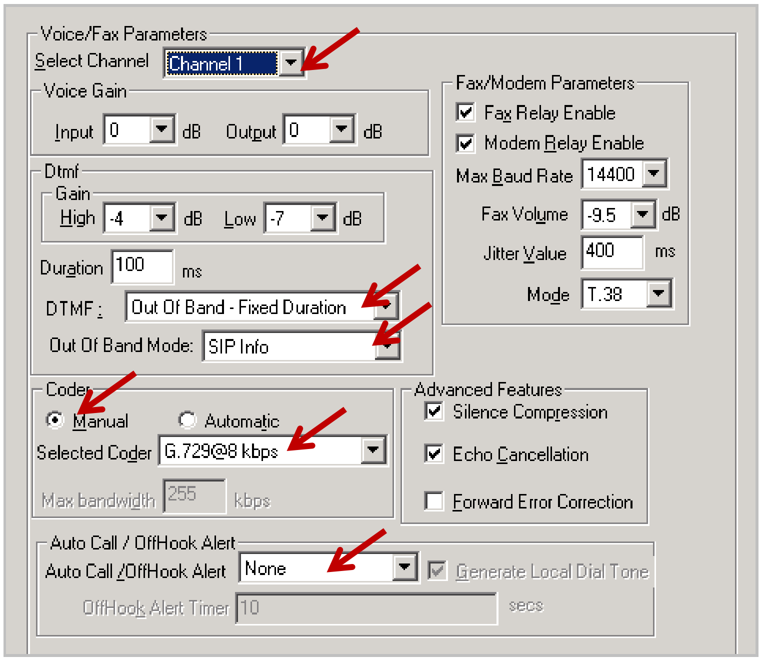
Follow these steps to configure the SIP extensions:

1. In MaxCS Administrator, select **Extension Configuration**.

2. Add two virtual extensions, 251 and 252, with the following parameters:
    - For *Type*, select **IP Extension**
    - Check **Enable IP Extension**
    - Select **Static IP Address**
    - Set *Static IP Address* to **10.10.101.81** (the IP address of the MultiVoIP gateway)

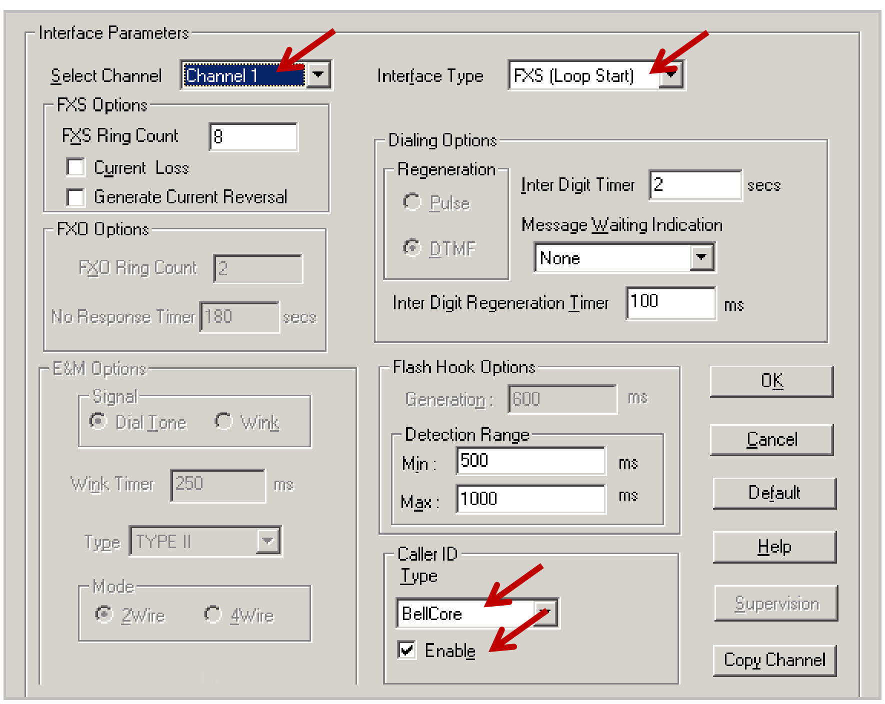


3. Open the MultiVoIP configuration tool.
4. Select **Configuration** > **Voice/Fax** and configure these parameters:
    - Set *Select Channel* to **Channel 01**
    - Set *DMTF to* **Out of Band - Fixed Duration**
    - Set *Out of Band Mode* to **SIP Info**
    - Set *Coder* to **Manual**
    - Set *Selected Coder* to **G.729@8kbps**
    - Set *Auto Call/OffHook Alert* to **None**
5. Click **Copy Channel** to apply the settings to all channels.



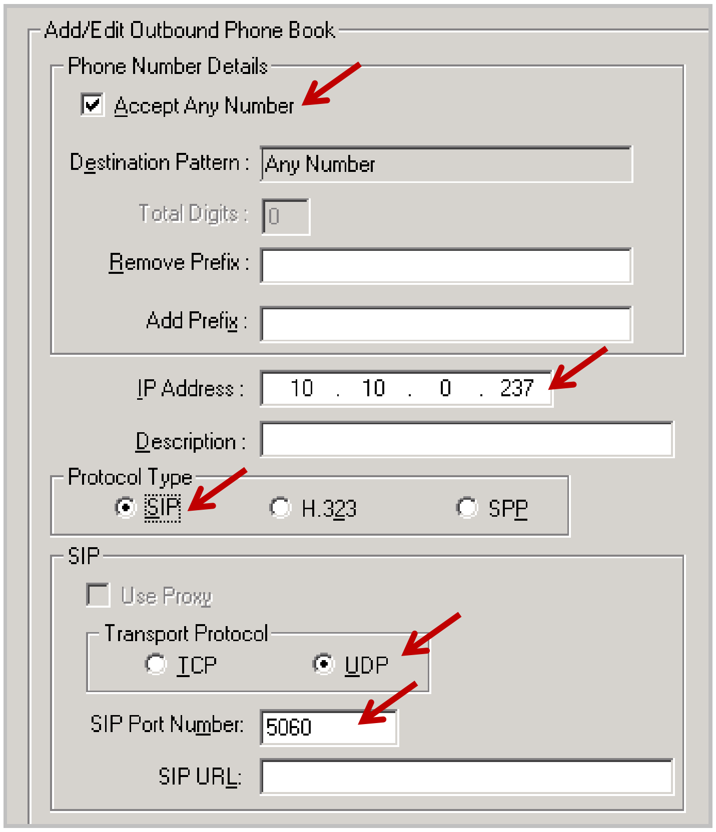
6. Select **Configuration** > **Interface** and configure these parameters:

- Set *Select Channel* to **Channel 1**
- Set *Interface Type* to **FXS (Loop Start)**
- Set *Caller ID Type* to **BellCore**
- Check the *Caller ID* **Enable** check box



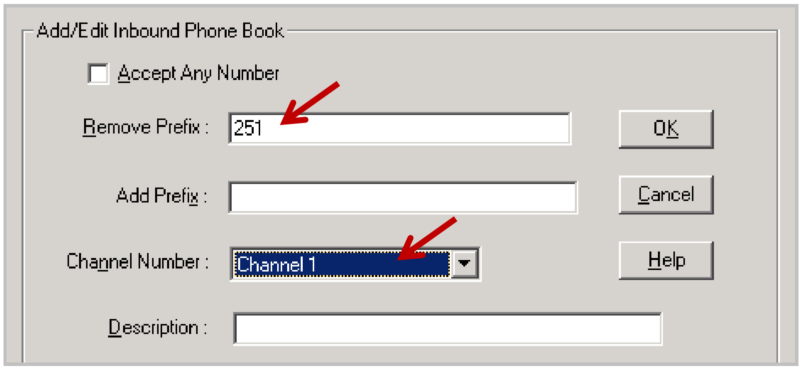
7. Select **Phone Book** > **Outbound Phone Book** and create an entry with the following parameters:

- Check **Accept Any Number**
- Set *IP address* to **10.10.0.237**(the MaxCS IP address)
- Set *Protocol* Type to **SIP**
- Set *SIP Transport Protocol* to **UDP**
- Set *SIP Port Number* to **5060**



8. Select **Phone Book** > **Inbound Phone Book** and create a new entry with these parameters:

- For *Remove Prefix* enter **251**
- Set *Channel Number* to **Channel 1** (this will bind Channel 1 to extension 251)



9. Repeat the same process for extension 252 by adding a new entry where *Remove Prefix* is **252** and *Channel Number* is **Channel 2**.
10. Under *Save & Reboot*, click **OK**. Wait for the MVP410 to reboot itself.
11. Attach analog phone sets to the MVP410 system (FXO/FXS port).
12. Make some test calls among ext 104, ext 251 and ext 252.

SIP-Tie Trunk Configuration for Incoming Calls from PSTN

Before you configure SIP-Tie trunks, check that all of the requirements in the Requirements section on page 3 have been met.

Analog extensions behind MVP410 are optional. If analog extensions are required, make sure you also complete the steps provided in the section SIP Extension Configuration.

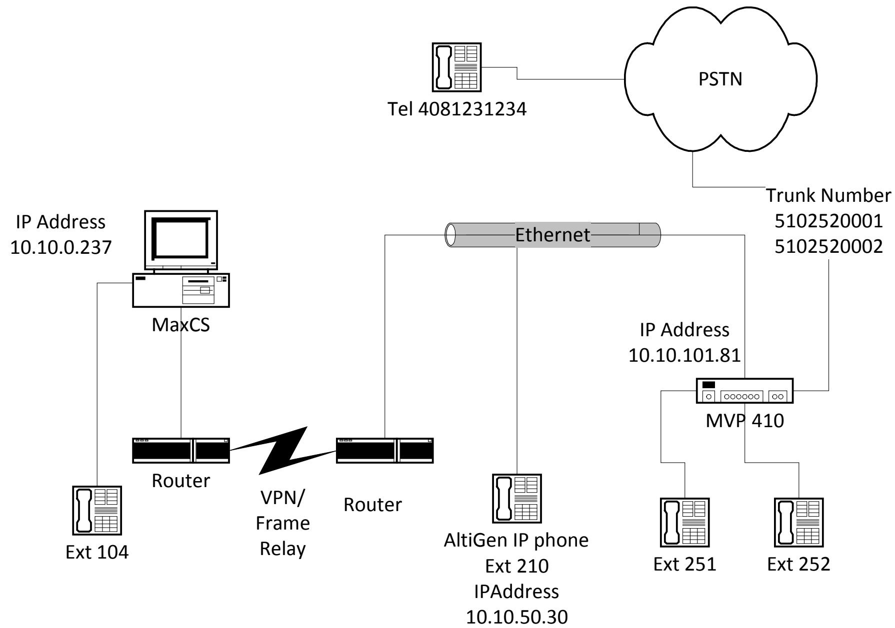


Figure 2: Configure SIP-Tie Trunks

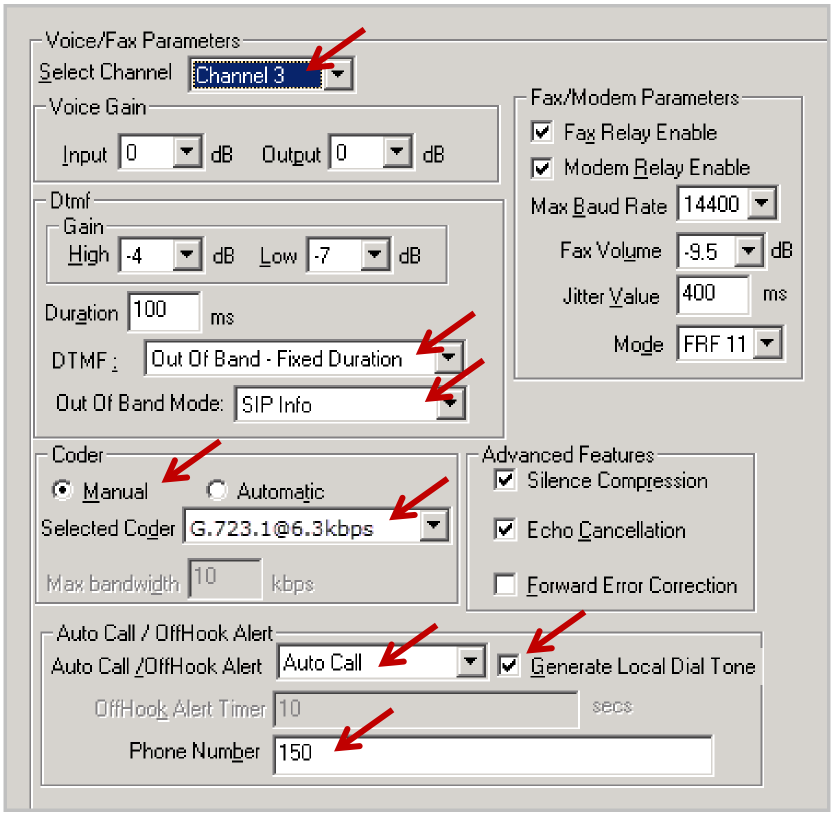
This section shows steps for the following example configuration:

- The MaxCS IP address at headquarters is 10.10.0.237
- MaxCS has one analog extension, extension 104
- At the remote site, the IP address of AltiGen IP phone extension 210 is 10.10.50.30
- The IP address of the MultiVoIP gateway is 10.10.101.81

Ext 251 and Ext 252 are optional; refer to SIP Extension Configuration for those instructions.

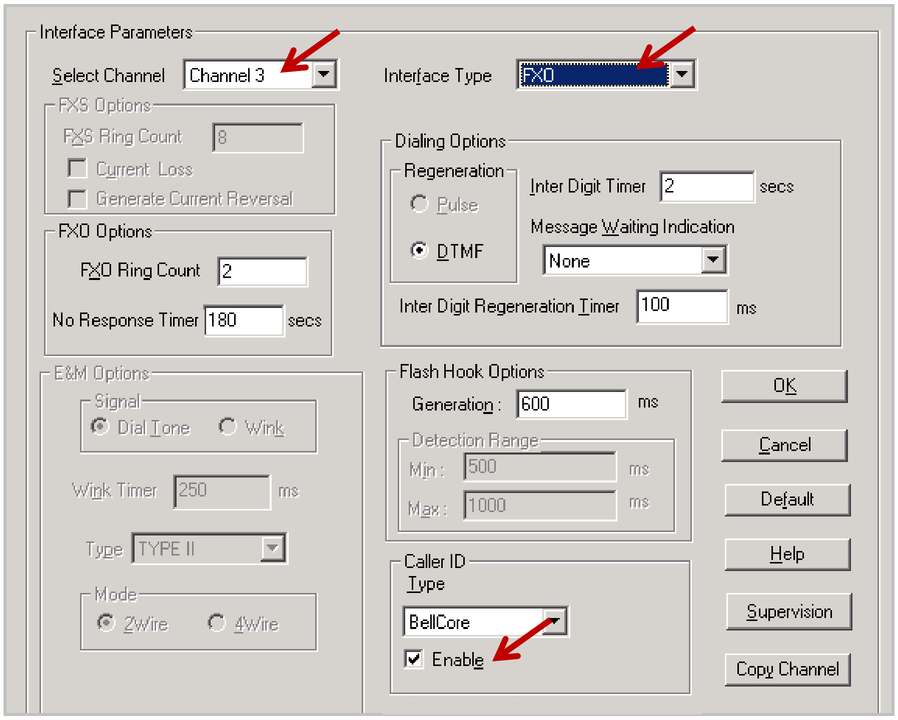
To configure SIP-Tie Trunks, follow these steps:

1. Open the MultiVoIP configuration tool.
2. Select **Configuration** > **Voice/Fax** and configure these parameters:
    - Set *Select Channel* to **Channel 3**
    - Set *DTMF* to **Out of Band – Fixed Duration**
    - Set *Out Of Band Mode* to **SIP Info**
    - Set *Coder* to **Manual**
    - Set *Selected Coder* to **G.723.1@6.3kbps**
    - Set *Auto Call/OffHook Alert* to **Auto Call**
    - Check *Generate Local Dial Tone*.
    - Set *Phone Number* to **150** (This means when the gateway sends a call to MaxCS, it will try to ring extension 150. If extension 150 does not exist, the call will be sent to the MaxCS Auto Attendant system.)



3. Select **Configuration** > **Interface** and configure the following parameters:

   - Set *Select Channel* to **Channel 3**
   - Set *Interface Type* to **FXO**
   - Set Caller ID *Type* to **BellCore**
   - Check the Caller ID *Enable* check box



4. Select **Configuration** > **Interface**.

5. Click the **Supervision** button and set the following parameters:

   - Check the **Answer Delay** checkbox
   - Set *Answer Delay Timer* to **1**

6. These settings only apply to Channel 3. Apply these setting to other FXO channels, if there are others.

7. Select **Save Setup** > **Save & Reboot**, click **OK**. Wait for the gateway to reboot itself.

Verification

1. Follow Figure 2 and attach the PSTN line 510-252-0001 to FXO/FXS Channel 3 behind the MVP gateway.

Important! Attach the line to FXS/FXO port instead of the E&M port.

2. Place a PSTN call from an outside line (for instance, from your cell phone) to 510-252-0001. The call should be sent to MaxCS through the MVP gateway.

3. Apply the same settings to Channel 4 on the MVP gateway. Attach the PSTN line 510-252-0002 to FXO/FXS Channel 4 behind the gateway.

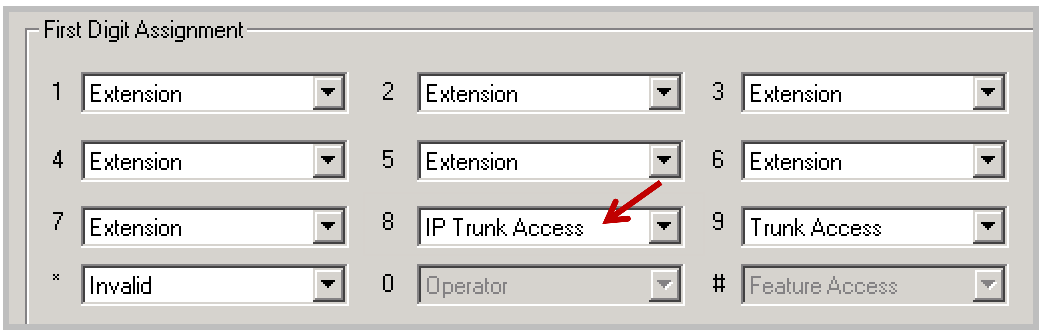
SIP-Tie Trunk Configuration for Outgoing Calls to PSTN

Before configuring these SIP-tie trunks, the requirements in the section Prerequisites must be met, and the steps in the section SIP-Tie Trunk Configuration for Incoming Calls from PSTN must to be completed.

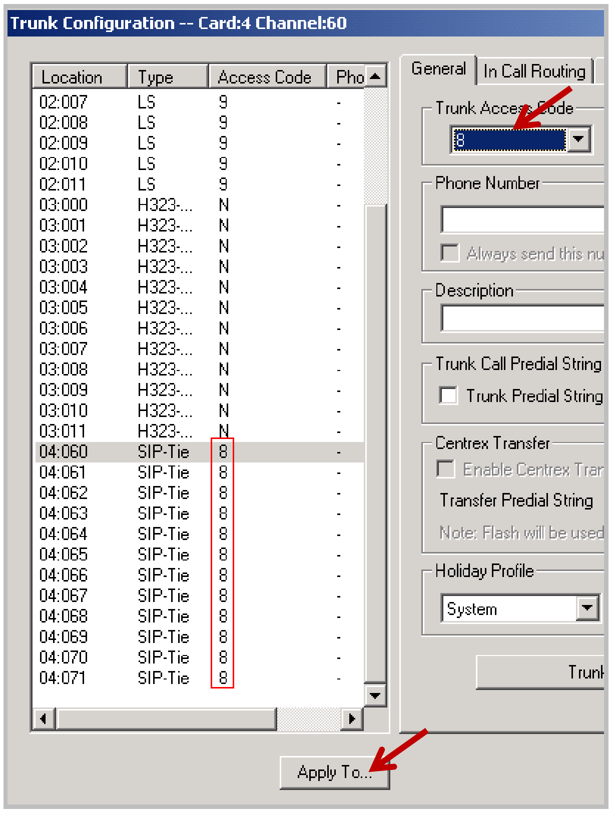
Analog extensions behind the MVP gateway are optional. If the analog extensions are required, complete the steps in the section SIP Extension Configuration before you proceed.

To configure the SIP-Tie Trunks, follow these steps:

1. Open MaxCS Administrator and select **System** > **System Configuration** > **First Digit Assignment**.
2. Assign digit 8 to **IP Trunk Access**.

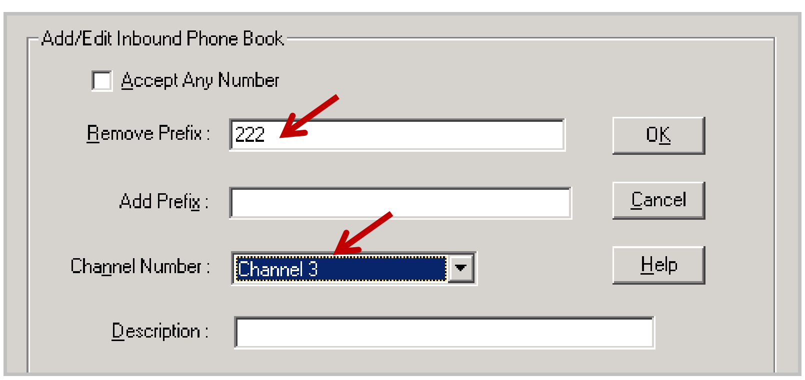


1. Select **PBX** >**Trunk Configuration** >**General**.
2. Highlight one of the SIP-Tie trunks.
3. Set *Trunk Access Code* to **8**.
4. Use the **Apply To** button to apply the settings to rest of the SIP-Tie trunks.



5. Log into the MultiVoIP tool.
6. Select **Phone Book** > **Inbound Phone Book** and add a new entry with the following parameters:
   - Set *Remove Prefix* to **222**

     Important! You can pick any prefix, but once the prefix is selected, it cannot be used as an extension number.
   - Set *Channel Number* to **Channel 3**
7. Repeat the same settings for FXO port Channel 4. (Apply the settings only for channels configured as FXO ports instead of FXS ports.)



8. Under **Save Setup** > **Save & Reboot**, click **OK**. Wait for the MVP410 to reboot itself.

Verification

1. Follow Figure 2, and attach a PSTN line 510252001 to FXO/FXS channel 3 behind the MVP gateway.

   Note: Attach the line to an FXS/FXO port instead of E&M port.

2. Pick up extension 104 (as illustrated in Figure 2) and dial "IP trunk access code" + "IP Dialing Table Entry ID" + "222" + "14081231234" + "#".

   In this example, the number to dial is 8 1 222 14081231234#.
   If "#" is not dialed, the call will still be sent out after a few seconds

Advanced Configuration

Before configuring these SIP-tie trunks, the requirements in the section Prerequisites must be met, and the steps in the sections SIP-Tie Trunk Configuration for Incoming Calls from PSTN and SIP-tie Trunk Configuration for Outgoing Calls to PSTN must be completed.

Analog extensions behind the MVP gateway are optional. If the analog extensions are required, complete the steps in the section SIP Extension Configuration before you proceed.

In addition, make sure you already can make calls among phones, the MVP gateway, and MaxCS without any problems.

Important! **FXS channels must be configured first.** For example, if your device has four channels and you want to provision two as analog extensions and the other two channels as SIP trunking channels, then you must configure channels 1 and 2 as FXS ports, leaving channels 3 and 4 for the trunks. If you do not provision the first two channels as the analog channels, then you may encounter one-way-only audio issues**.**

Outgoing Calls to PSTN Using Out Call Routing (Optional)

To make a call to the outside though the MVP gateway requires dialing a number with a "strange" prefix, which may not be intuitive for most users.

For example, users may expect to dial "914081231234" instead of "8-1-222-14081231234."

Proper configuration of out-call routing in MaxCS can resolve this problem. For more information on configuring out-call routing, refer to the *Out Call Routing* chapter in the *MaxCS Administration Manual* or search for "routing" in the MaxCS online Help system.

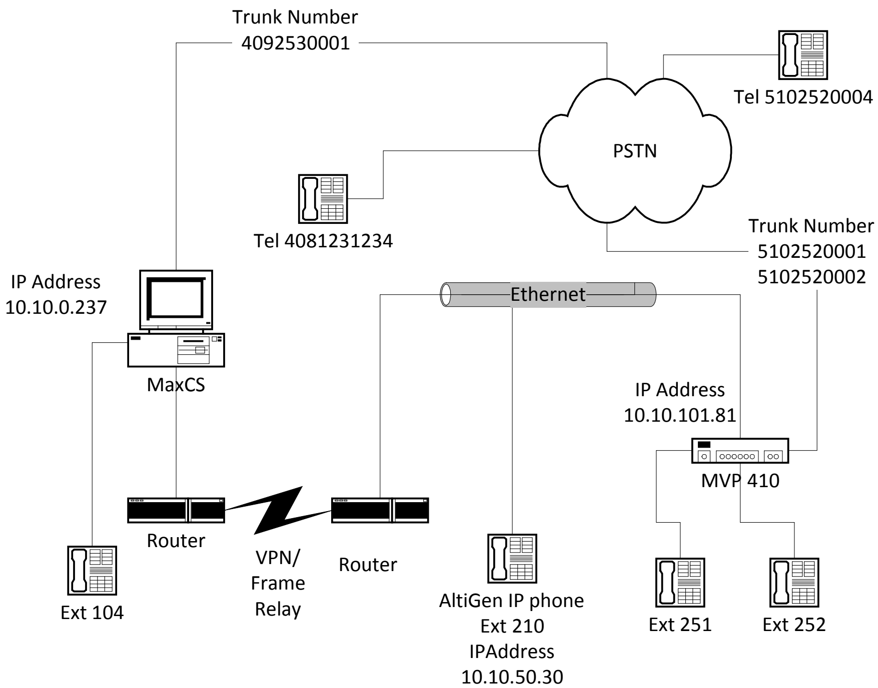


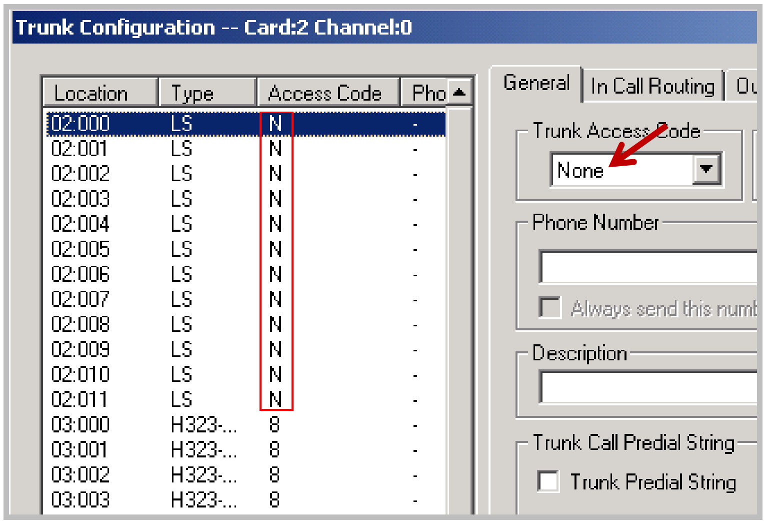
Figure 3: Using Out Call Routing

This section shows steps for the following example configuration:

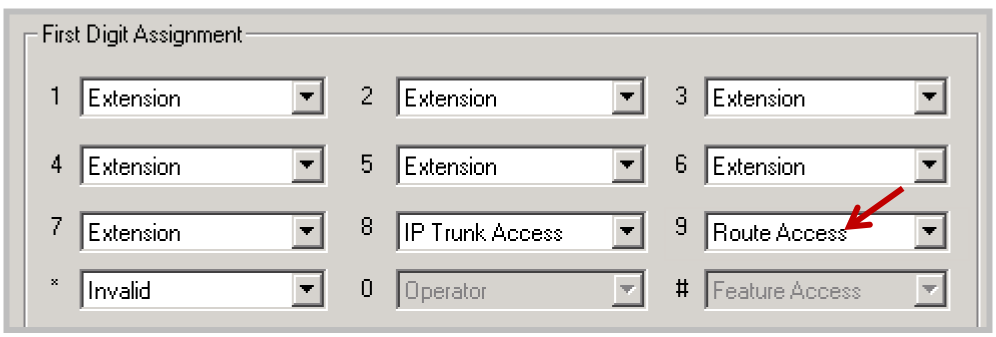
- The IP address of MaxCS at headquarters is 10.10.0.237
- MaxCS has one analog extension, extension 104
- The home area code of the MaxCS system is 409
- The MaxCS system has several analog trunks attached where the number is 4092530001.
- In the remote site, the IP address of the MVP gateway is 10.10.101.81
- Two trunks are attached to MVP gateway FXO ports where the numbers are 5102520001 and 5102520002

To configure outgoing PSTN call via Out-call routing, follow these steps:

1. Log into MaxCS Administrator and select **PBX** > **Trunk Configuration**.
2. Set all analog trunk *Access Codes* to **None** (the H.323-Tie or SIP-Tie trunk access code should still be “8”).

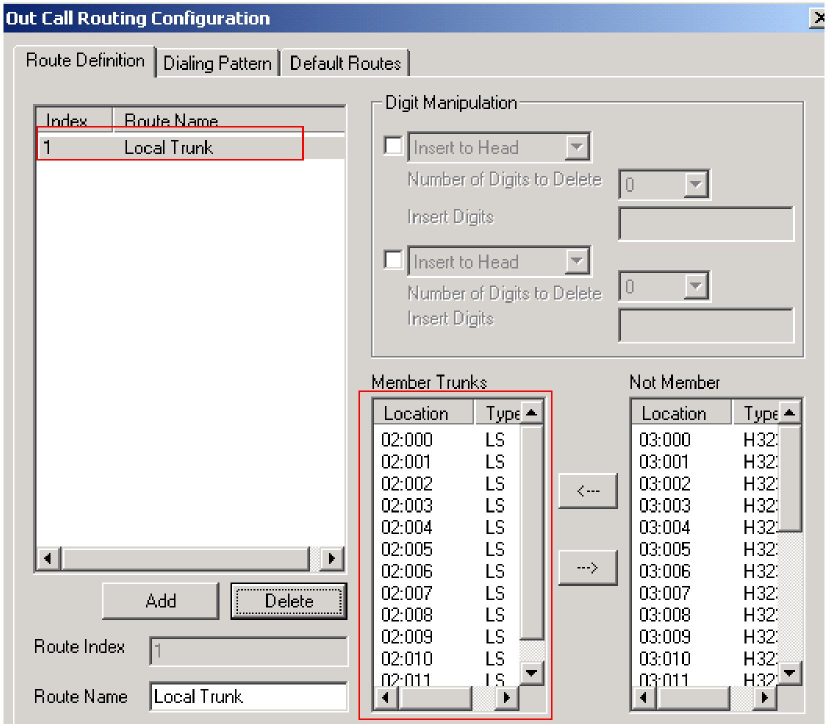


3. Select **System** > **System Configuration** > **First Digit Assignment**.
4. Assign digit 9 to **Route Access**.

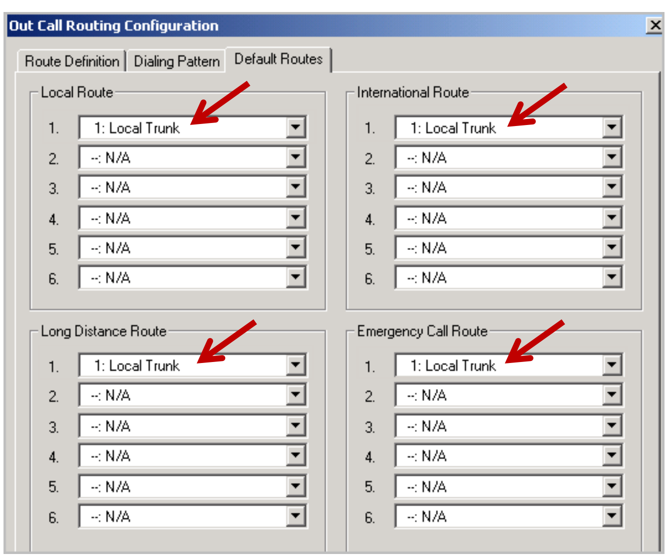


5. Select **PBX** > **Out Call Routing Configuration** > **Route Definition**.
6. Add a new entry named **Local Trunk**.

7. Select all the analog trunks (in this example, from “02:000” to “02:011”) as **Member Trunks.**
8. Click **Apply**.



9. Select **PBX** > **Out Call Routing Configuration** > **Default Routes** and configure these parameters:
   - Set *Local Route* entry 1 to **1: Local Trunk**
   - Set *Long Distance Route* entry 1 to **1: Local Trunk**
   - Set *International Route* entry 1to **1: Local Trunk**
   - Set *Emergency Call Route* entry 1to **1: Local Trunk**
10. Click **Apply**.



Verification

At this point, you should be able pick up extension 104 and make an outbound call by dialing 914081231234.

Outpost Digits

In Figure 3, when a user makes a call to 15102520004 from extension 104, it would be nice if the system would send the call through the MVP gateway's FXO port channel 3 or channel 4.

To achieve this, MaxCS needs to outpost the digits "2222520004" to the gateway. To do this, MaxCS must remove the first 4 digits of "15102520004", and then insert "1222" to the head.

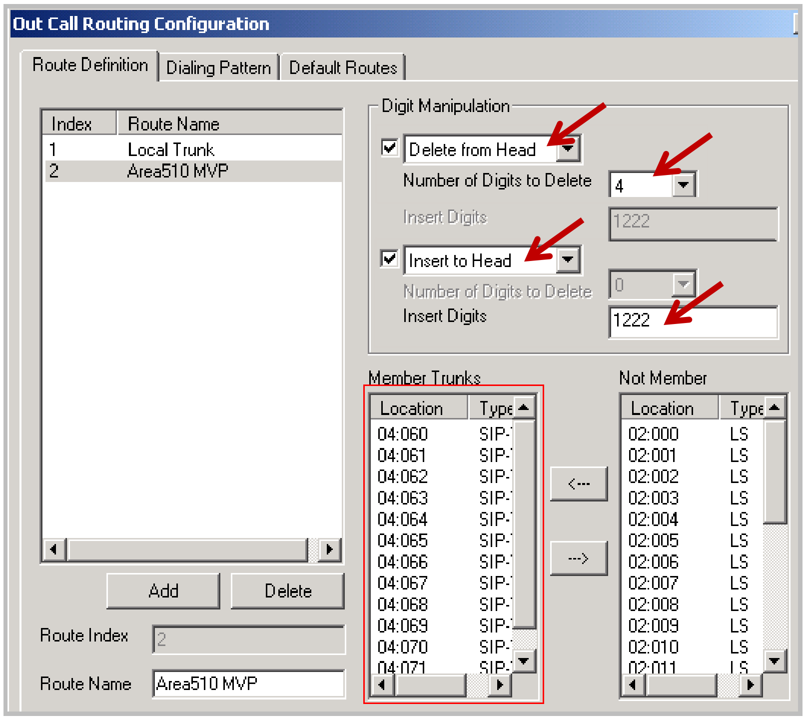
"1" of "1222" is the Location ID in the IP dialing table.

The final number sent to the gateway would be "2222520004."

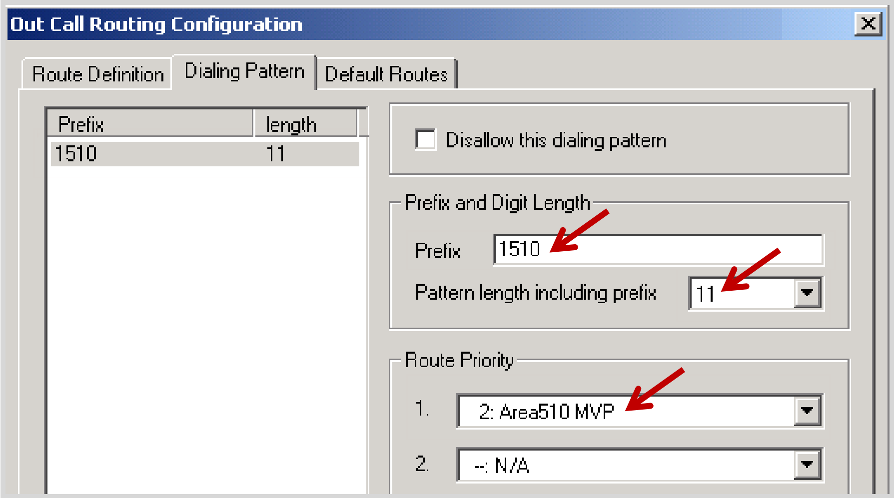
The gateway will remove the "222" from "2222520004" and then outpost "2520004" to the FXO trunk.

To configure this digit processing, follow these steps:

1. Log into MaxCS Administrator and select **PBX** > **Out Call Routing Configuration** > **Route Definition**.
2. Add a new entry named **Area510MVP** with the following parameters:
    - In the Digit Manipulation panel, check **Delete from Head**
    - Set *Number of Digits to Delete* to **4**
    - Check **Insert to Head**
    - Set *Insert Digits* to **1222**
3. Set all SIP-Tie trunks (in this example, from "04:060" to "04:071") as **Member Trunks**. Click **Apply**.



4. Select **PBX** > **Out Call Routing Configuration** > **Dialing Pattern** and add a new entry:
   - Set *Prefix* to **1510**
   - Set *Pattern length including prefix* to **11**
   - Set *Route Priority 1* to **2: Area510 MVP**
5. Click **Apply**.



Verification

1. Follow Figure 3. Use extension 104 to dial 15102520004. The call should go through the MVP gateway.
2. Use extension 104 to dial 14081231234. The call should go through local trunks instead of the gateway.

Outgoing Call to PSTN Using Extension Dialed Digit Translator

Before you start these procedures, make sure you have met the requirements in the section Prerequisites, and perform the steps in the sections SIP-tie Trunk Configuration for Incoming Calls from PSTN and SIP-tie Trunk Configuration for Outgoing Call to PSTN. The steps in the section Outgoing Calls to PSTN using Out Call Routing are optional. Analog extensions behind MVP410 are optional. If analog extensions are required, also perform the steps in the section SIP Extension Configuration.

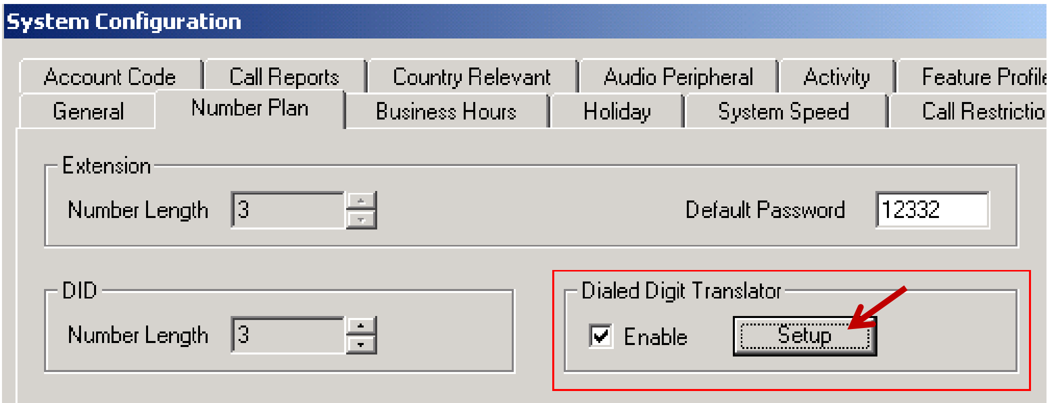
The MultiVoIP gateway is typically located in a remote branch office. When a user in the remote office wants to make a trunk call, for voice quality purposes the remote user may always want the call go through the MVP410 FXO ports. An administrator can configure “Extension Dialed Digit Translator” to achieve this goal.

Please refer to Figure 3. When a remote user uses extension 251 or 210 to make a call to 15102520004, MaxCS needs to translate the number to 8-1-222-2520004 (Refer to the section SIP-tie Trunk Configuration for Outgoing Calls to PSTN for details).

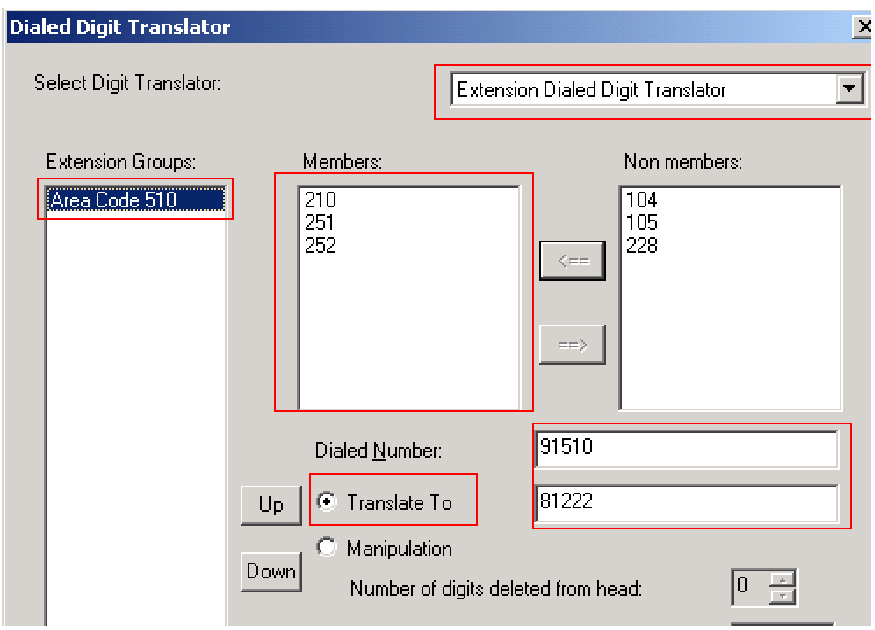
(Ext 251 and ext 252 are optional if you only use AltiGen IP phone 210.)

To configure this digit translation, follow these steps:

1. Log into MaxCS Administrator and select **System** > **System Configuration** > **Number Plan**.
2. In the *Dialed Digit Translator* panel, check the **Enable** check box and click **Setup**.



1. In the Dialed Digit Translator window, from the *Select Digit Translator* list select **Extension Dialed Digit Translator**.
2. Create a new Extension group named **Area Code 510**.
3. Move extensions **210**, **251** and **252** to the **Members** list. (Extensions 251 and 252 are optional if you only have AltiGen IP phone 210.)
   - For *Dialed Number* enter **91510**
   - For *Translate To* enter **81222**
4. Click **Apply**.



Verification

From extension 210 and extension 251 in the remote site, make a call to 9-1-510-250004. MaxCS should send the call through the MultiVoIP gateway's FXO ports.

Note: With this approach, the user will always need to dial 12 digits, even when placing a local call. (In this example, the user must dial 9-1-510-252-0004 instead of dialing 252-0004.)

"911" Using Extension Dialed Digit Translator

Before you start these procedures, make sure you have met the requirements in the section Prerequisites, and perform the steps in the sections SIP-tie Trunk Configuration for Incoming Calls from PSTN and SIP-tie Trunk Configuration for Outgoing Call to PSTN. The steps in the section Outgoing Calls to PSTN using Out Call Routing are optional. Analog extensions behind MVP410 are optional. If analog extensions are required, also perform the steps in the section SIP Extension Configuration.

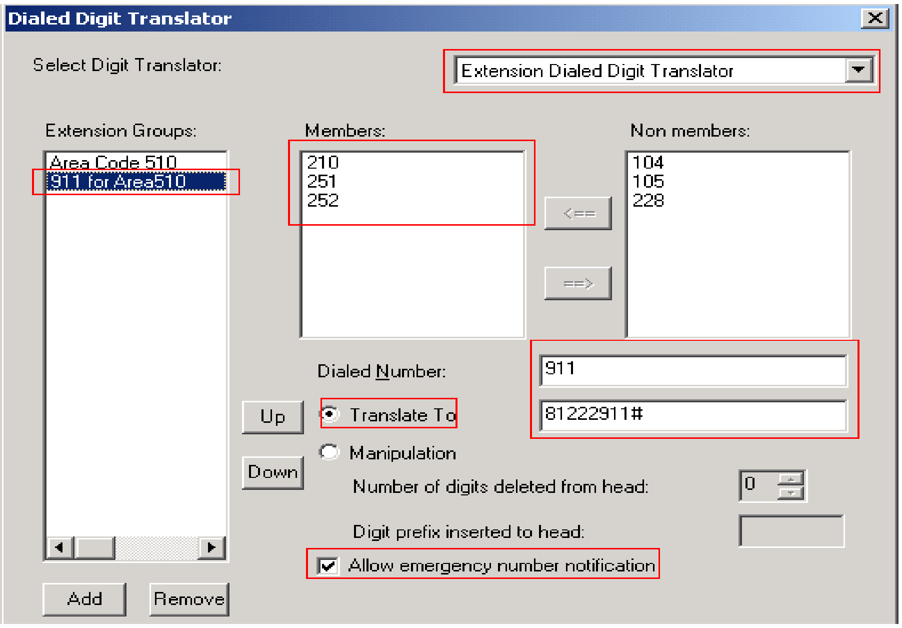
The MultiVoIP gateway is typically located in a remote branch office. When a user in the remote office wants to make a 911 emergency call, the call should always go through the MVP gateway's FXO ports. Otherwise, the call might reach the wrong 911 center.

For example, a user in San Francisco dials 911, but the call is sent to a New York 91" center. An administrator can configure the *Extension Dialed Digit Translator* to make sure that 911 calls are always sent to the correct 911 center.

To configure 911 digit translation, follow these steps:

Under Dialed Digit Translator:

1. Log into MaxCS Administrator and select **System** > **System Configuration** > **Number Plan**.
2. From the *Select Digit Translator* list, select **Extension Dialed Digit Translator**.
3. Create a new Extension Group named **911 for Area510**.



4. Move extensions **210**, **251**, and **252** to the **Members** list. (Extensions 251 and 252 are optional if you only have AltiGen IP phone 210.)
    - For *Dialed Number* enter **911**
    - For *Translate To* enter 81222911# (The # symbol will eliminate the time out)
5. Click **Apply**.

Verification

From extensions 210 and 251 in the remote site, dial 911. MaxCS should seize the MultiVoIP gateway's FXO port to make 911 calls.

AltiGen IP Phone Emergency GW Configuration

Before you begin these procedures, make sure you already met the Prerequisites and performed steps in the SIP-tie Trunk Configuration for Incoming Calls from PSTN section and the SIP-tie Trunk Configuration for Outgoing Calls to PSTN section.

Refer to Figure 3. When the intranet between headquarters and the remote office is down, IP phone 210 will change to Basic mode. Through proper configuration, an AltiGen IP phone user still can dial 911.

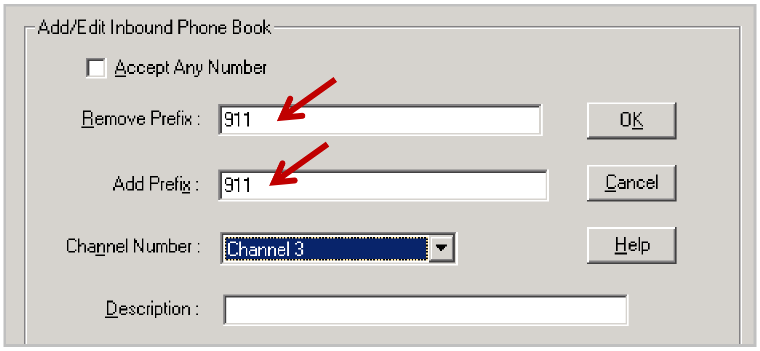
To configure emergency gateway dialing, follow these steps:

1. Log into the MultiVoIP configuration tool and select **Phone Book** > **Inbound Phone Book**.
2. Add a new entry.
    - Set both *Remove Prefix* and *Add Prefix* to **911**

Note: If you want to do some tests without really sending the call to the 911 center, you can temporarily change this value to, for example, your cell phone number.

   - Set *Channel Number* to **Channel 3**

3. Repeat the same settings for FXO port Channel 4. (Apply the settings only to channels configured as FXO ports instead of FXS ports.)



4. Select **Save Setup** > **Save & Reboot**, click **OK**. Wait for the gateway to reboot itself.
5. On AltiGen IP phone extension 210, press **#26** and press **Enter** to log out of the extension. It will change to Basic mode. On the AltiGen IP phone, press the **Menu** button. (For the AltiGen IP 600 phone, press ****7** and press **Enter**.)

6. On the AltiGen IP phone, set the following parameters:
    - Select **Network** > **Enable NAT** and set this to No.
    - Select **System** > **Emergency Num** and set this to **911**
    - Select **System** > **Emergency GW** and set this to **10.10.101.81** (the MultiVoIP gateway's IP address)

Verification

1. Make sure that IP phone extension 210 is in Basic mode.
2. On IP phone extension 210, dial **911.** The call should be sent to the 911 center or to the test number you specified in step 2 of the previous section for *Add Prefix*. (Found in **Phone Book** > **Inbound Phone Book**)
3. Refer to Figure 3 on IP phone extension 210, and dial 1-408-123-1234. The phone 408-123-1234 should ring within 10 seconds.
4. Now you can log in IP phone 210 to extension 210 by pressing **#27**.

Operational Limitations

When configuring and using this device, be aware of the following limitations.

1. The FXO ports cannot support DTMF.
2. A special firmware version, version 6.09.16-04-Apr-07-Altigen, is supported. This firmware version is old (April 2007), and may not work well with the MVP devices that have just been manufactured.
3. Internet Explorer version 6.0 is required to fully support the web interface configuration. However, this version of IE is not supported anymore, so the device can only be configured using an RS232 connection.
4. Caller ID does not work with an FXO configuration.

AltiGen Technical Support

AltiGen does not provide general configuration support for EdgeWater or AudioCodes products. For general configuration information, refer to your AudioCodes or EdgeMarc documentation.

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