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Time Warner ITSP Setup Guide

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This configuration guide was created to assist knowledgeable vendors with configuring the Zultys MX Phone System with Time Warner SIP Trunks using Innomedia ESBC 9378-4B Ver. 2.0.12.42-OR, Innomedia ESBC 10K-MDX Ver. 2.0.13.0-OR ITSP service.

The Zultys MX must be running 9.0.4 for proper operation.

Questions about software installation or other PBX configuration issues should be directed to Zultys support at support@zultys.com. The Zultys Phone System Manual is available on the Zultys KBS: <http://kbs.zultys.com>.

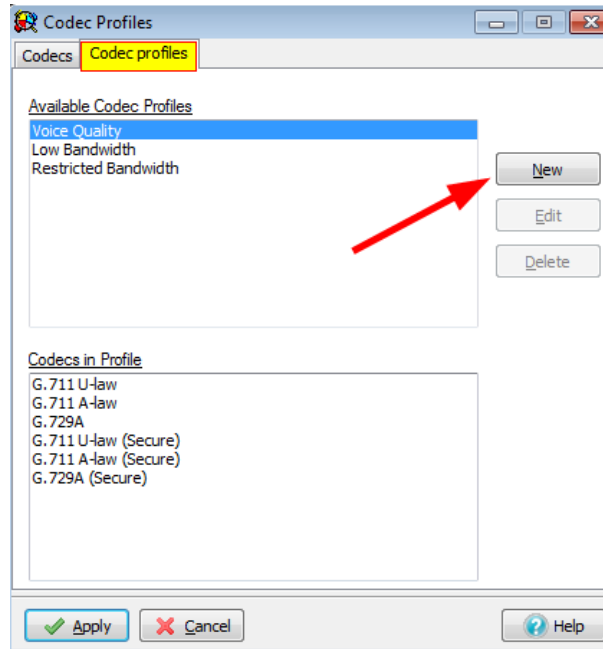
NOTE: in this document all IP addresses are example IP addresses, please replace with a valid IP of your MX

Content

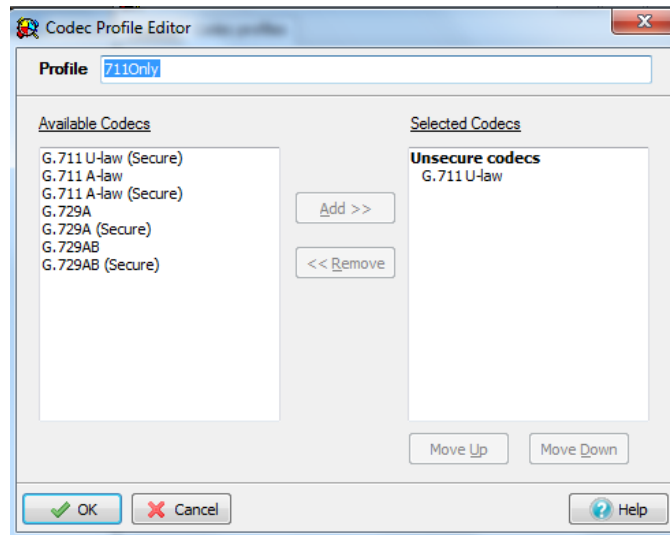
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I. Create Custom Codec Profile

This carrier only supports G711 which requires the MX to only offer G711 in its SDP packets. This will require a new custom codec profile to be created. In the MX Administration click Provision | Codec, then click on the Codec profiles tab, and then click the New button.



In the resulting Pop Up, provide a descriptive name for the new codec profile, such as 711Only, and from the list of available codec select ONLY G.711 U-law, and click OK.



The Codec Profile Editor, as shown above, creates and edits Codec Profiles. You access the Codec Profile Editor by pressing the New or Edit button in the Codec Profiles window.

1.1 Panel Contents

The Profile data entry box at the top of the panel lists the name of the codec profile that you are editing. G711 is the name entered in this field. The Available Codecs table on the left side of the panel lists the codecs that are available and are not included in the current codec

profile. The Selected Codecs table on the right side of the panel lists the codecs that are included in the current codec profile.

1.2 Creating or Editing a Profile

To create a profile, enter the name of a new profile in the Profile data entry box at the top of the panel. If you enter this panel by pressing the New button in the Codec Profiles window, this field is blank until you enter a name.

To edit a profile, enter the name of an existing profile in the Profile data entry box. If you enter this panel by pressing the Edit button in the Codec Profiles window, the name of the highlighted profile in that panel is placed in the Profile data entry box.

1.3 Modifying Profile Parameters

To add codecs to the current profile, select one or more codecs in the Available Codecs table with your mouse cursor and press the Add button between the tables.

To remove codecs from the current profile, select one or more codecs in the Selected Codecs table with your mouse cursor and press the Remove button between the tables.

To change the order in which codecs are presented to other SIP devices when negotiating communication parameters, select the codec in the Selected Codecs table and press the Move Up or Move Down buttons below the table.

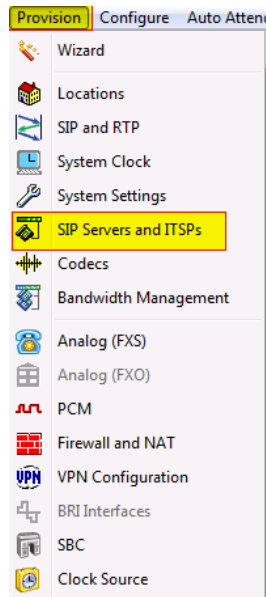
To rename a profile, enter the new name for the profile in the Profile data entry box at the top of the panel.

1.3.1 Button Bar

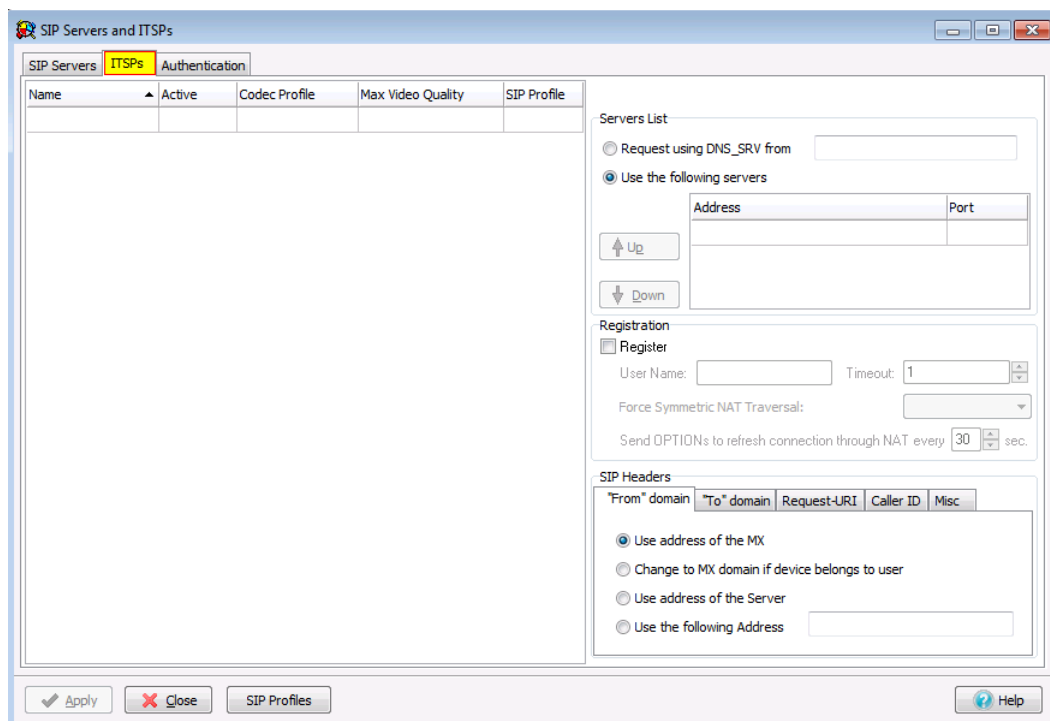
- **OK:** Pressing the OK button closes the Codec Profile Editor and places the changes in the Codec Profiles window. These changes are then permanently saved to the database only after you press the Apply button in the Codec Profiles window. An inactive OK button (button contents are grey) indicates that you have not altered the selected profile.
- **Cancel:** Pressing this button discards changes made within the data entry form and returns you to the Codec Profiles window.

2. Create ITSP Account

In the MX Administrator, go to *Provision* then *SIP Servers and ITSPs*.



This will bring up the SIP and ITSP configuration screen. It has three tabs along the top. To configure the SIP server, click on the *SIP* tab. For ITSPs, click on the *ITSP* tab. To set up the authentications, click on the last tab, *Authentication*.



The SIP Servers and ITSPs tabs are both divided into two sections:

- **The Servers Table**, located on the left, lists the servers that provide voice session access to MX devices.
- **The Properties Table**, located on the right, configures the address used to access the servers and specifies transmission characteristics of SIP packets that set up the voice sessions.

2.1 Servers Table

The Servers table in the *SIP Servers and ITSPs* tab lists the SIP servers accessed by the MX to establish voice call sessions. Each row corresponds to a SIP server. The following parameters identify the characteristics of each SIP server.

- **Name:** This parameter identifies the SIP Server to the MX. Other UI windows, such as the Dial Plan: Routing panel, reference SIP Servers by their names.
- **Active:** This parameter specifies the active status between the MX and the SIP server. If this parameter is not selected, the MX cannot use the specified SIP server to route a call.
- **Type:** This parameter specifies how incoming calls from the SIP server are handled by the MX. Valid parameter settings include:
 - **Internal:** The number specified in the SIP INVITE is treated as a dialing pattern that is evaluated by the Routing panel of the Dial Plan window.
 - **External:** The number specified in the SIP INVITE is treated as a DID and routed to the user that is assigned to that number. Calls with unrecognized DID numbers are handled as specified by the Outside panel of the Dial Plan. All servers in the ITSP panel are external. This parameter is not listed in the Servers table of the ITSP panel.
- **Codec Profile:** Specifies the list of codecs that the SIP server can use for negotiating communication settings with other SIP devices. Codec Profiles configured in your system are listed in the Codec Profiles window.
- **SIP Profile:** SIP profiles define SIP packet characteristics for packets utilizing the specified SIP server. Click the *SIP Profiles* button located at the bottom of the panel for a list of SIP Profiles and their definitions.

To add a SIP Server to the table, right-click the mouse while pointing at the table and select *New*. Enter the server parameters in the new row.

To edit an existing SIP Server, double-click in the appropriate cell and enter the new information.

To remove a SIP Server from the table, select the server, right-click the mouse, and select *Delete* from the menu.

2.2 Properties Table

The properties Table defines connection, registration, and SIP packet characteristics for the SIP server highlighted in the Servers Table. The text at the top of the table, above the Servers List, identifies the server configured by the Properties Table.

- **Servers List:** This table section defines the access address of the selected SIP Server:
 - **Request using DNS_SRV:** Select this option to specify an FQDN that is associated with the SIP server. The MX uses the DNS server to resolve the IP address and port of the server.
 - **Use the following servers:** Select this option to specify one or more SIP Server addresses (using dotted decimal notation or FQDN) and port number configurations through which the MX performs voice calls. To add server addresses to the table, point the cursor at the table and right-click the mouse.
 - **Registration:** This section specifies the registration parameters that allow the MX to register as a client to the selected SIP server.
 - **Register:** Check this box to enable the MX to register as a client to the specified SIP Server.
 - **User Name:** If the Registration option is enabled, this parameter specifies the string that is designated as the user name in the From field for INVITE packets sent from the MX to the SIP Server. The From field derives the Domain name on the basis of the Domain in From Header parameter.
 - **Timeout:** If the Registration option is enabled, this parameter specifies the registration period for the MX.
 - **Force Symmetric NAT Transversal:** This field regulates how RFC3581 "An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing" is used. There are three possible options for this field:
 - **Off:** RFC3581 defined behavior is disabled.
 - **On:** The MX always enforces RFC3581 when communicating to this ITSP. OPTIONS SIP message is used to maintain NAT binding.
 - **Auto:** The MX attempts to determine via rport parameter if it is behind the NAT and if so enforces NAT binding using OPTIONS SIP message. With the introduction of SBC in version 5.0.15 software and later, it is recommended not to use this configuration option.
 - **Domain in "From" Header:** For INVITE messages that are sent from the MX through the SIP Server, this parameter specifies the display name and URL that is placed in the From Header:

- Select *Use Address of the MX* to designate the MX as the originator address.
 - Select *Change to MX Domain* if device belongs to a user, to specify the MX as the originator address when the MX receives the message from an MX User. If the message is received from an unknown user (for example, a message may be received from an external source through the SIP server), the MX does not alter the From header.
 - Select *Use Address of the Server* to specify the SIP server as the originator address.
 - Select *Use the Following Address* and then enter an IP address, to specify another unrelated address as the originator address.
- **Domain in "To" Header:** For INVITE messages that are sent from the MX through the SIP Server, this parameter specifies the display name and URL that is placed in the To Header:
- Select *Use the domain name for the appropriate server in the Servers List* to designate the domain name of the ITSP as the destination address.
 - Select *Use resolved address for the appropriate server in the Servers List* to designate the IP address resolved from the domain name of the ITSP as the destination address.
 - Select *Use custom domain name* and then enter a domain name, to specify another unrelated domain name as the destination address.
- **Request-URI:** For INVITE messages that are sent from the MX through the SIP Server, this parameter specifies the Request URI that is placed in the Request-URI:
- Select *FQDN* to designate the domain name of the MX as the Request-URI address.
 - Select *IP Address* to designate the IP address of the MX as the Request-URI address.
 - Select *Use custom domain name* and then enter a domain name, to specify another unrelated domain name as the Request-URI address.
- **Caller ID:** Sets caller ID options provided by the MX
- FROM Header
 - Registration User Name: Populate the Caller ID as the Registration User Name
 - Originating party Caller ID: Populate Caller ID as the Originating Party Caller ID

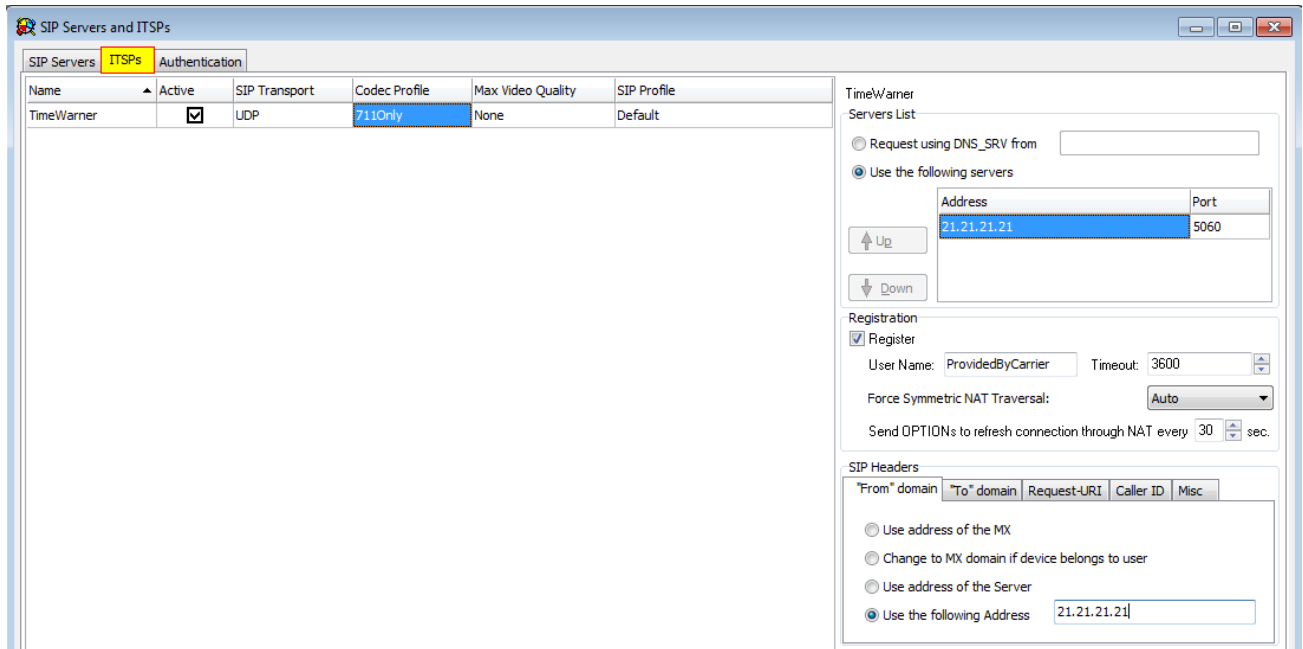
- Custom: Populate Caller ID with the custom value entered
- Include optional identity header: Select
 - P-Asserted-Identity: to populate the P-Asserted Identity field with the following property
 - Registration User Name: Populate the Caller ID as the Registration User Name
 - Originating party Caller ID: Populate Caller ID as the Originating Party Caller ID
 - Custom: Populate Caller ID with the custom value entered
 - Remote Party ID: to populate the Remote Party ID field with the following property
 - Registration User Name: Populate the Caller ID as the Registration User Name
 - Originating party Caller ID: Populate Caller ID as the Originating Party Caller ID
 - Custom: Populate Caller ID with the custom value entered
- **Misc:** Contains miscellaneous settings
 - Select *Ignore Domain in From/To SIP headers* to have the MX ignore the domain received from the ITSP on all packets.
 - Select unsupported media type response code to send to the ITSP when an unsupported media type is received from the ITSP. Options are 415 or 488

3. Configure the account

Enter the following information:

- **Name:** TimeWarner
- **Active:** Check
- **Codec Profile:** Select *711 Only (Customer profile created earlier)*
- **SIP Profile:** Select *Default*
- **Video Codec:** Select *None*
- **Use the following servers:** Provided by the Carrier
- **Port:** 5060
- **Register:** Checked
- **Username:** Provided by the Carrier

- **Force Symmetric NAT Traversal:** Auto
- **From header:** Select *Use the following Address: IP Provided by Carrier*
- **To header:** Select *Use custom domain name: IP Provided by Carrier*
- **Request-URI:** Select *IP Address*
- **Caller ID:** Select *Registration User Name*
 - An alternate method is described in Alternate Caller ID Setup below where *Custom* set to one of their DID's
- **Misc:** Nothing selected



Name	Active	SIP Transport	Codec Profile	Max Video Quality	SIP Profile
TimeWarner	<input checked="" type="checkbox"/>	UDP	711Only	None	Default

TimeWarner

Servers List

Request using DNS_SRV from []

Use the following servers

Address	Port
21.21.21.21	5060

Registration

Register

User Name: ProvidedByCarrier Timeout: 3600

Force Symmetric NAT Traversal: Auto

Send OPTIONS to refresh connection through NAT every 30 sec.

SIP Headers

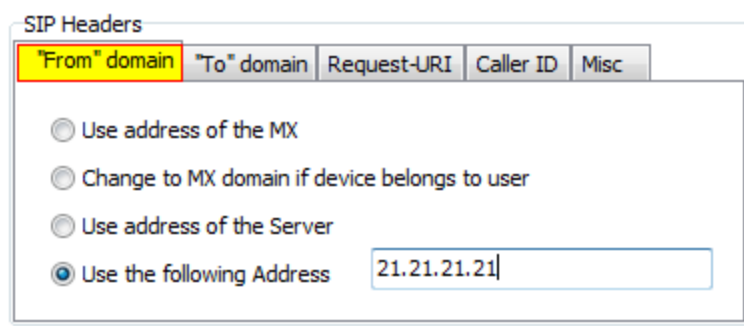
"From" domain | "To" domain | Request-URI | Caller ID | Misc

Use address of the MX

Change to MX domain if device belongs to user

Use address of the Server

Use the following Address [21.21.21.21]



SIP Headers

"From" domain | "To" domain | Request-URI | Caller ID | Misc

Use address of the MX

Change to MX domain if device belongs to user

Use address of the Server

Use the following Address [21.21.21.21]

SIP Headers

"From" domain "To" domain Request-URI Caller ID Misc

Use the domain name for the appropriate server in the Servers List
 Use resolved IP address for the appropriate server in the Servers List
 Use custom domain name

SIP Headers

"From" domain "To" domain Request-URI Caller ID Misc

FQDN
 IP Address
 Use custom domain name

SIP Headers

"From" domain "To" domain Request-URI Caller ID Misc

FROM Header

Include optional identity header

SIP Headers

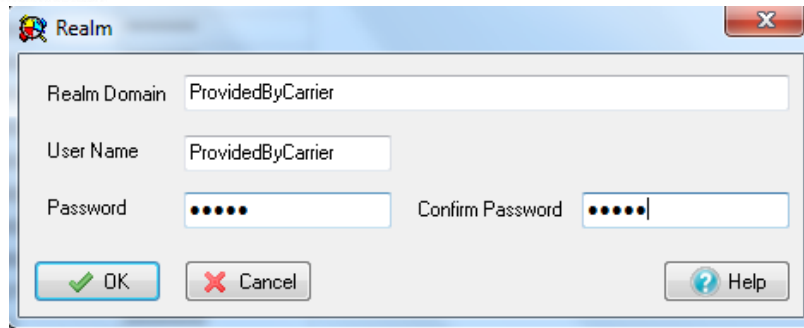
"From" domain "To" domain Request-URI Caller ID Misc

Ignore Domain received in From/To SIP headers

Unsupported media type response code

4. Authentication

The authentication tab is used to define the authentication information for realm based authentication.



- **Realm Domain:** Provided by Carrier
- **User Name:** Provided by Carrier
- **Password:** Provided by Carrier

4.1 Table Parameters

Each row corresponds to a Protection Domain that the MX is allowed to access.

Realm: This parameter specifies the domain name of the realm. It must match the domain that is offering the challenge.

User Name: This parameter specifies the User Name under which the MX is allowed to access the realm.

Password: This parameter specifies the password that the MX uses to authenticate itself with the realm.

4.2 Editing the Table

To add to a realm, access the Realm panel by right-clicking in the table and selecting Add.

To edit a realm, access the Realm panel by double-clicking in one of the Realm entry's cells or select a Realm, right-click the mouse, and select Edit.

To remove a realm from the table, select the realm, right-click the mouse, and select Delete from the menu.

5. Enabling ALG / SBC on MX (MX version 5.0+)

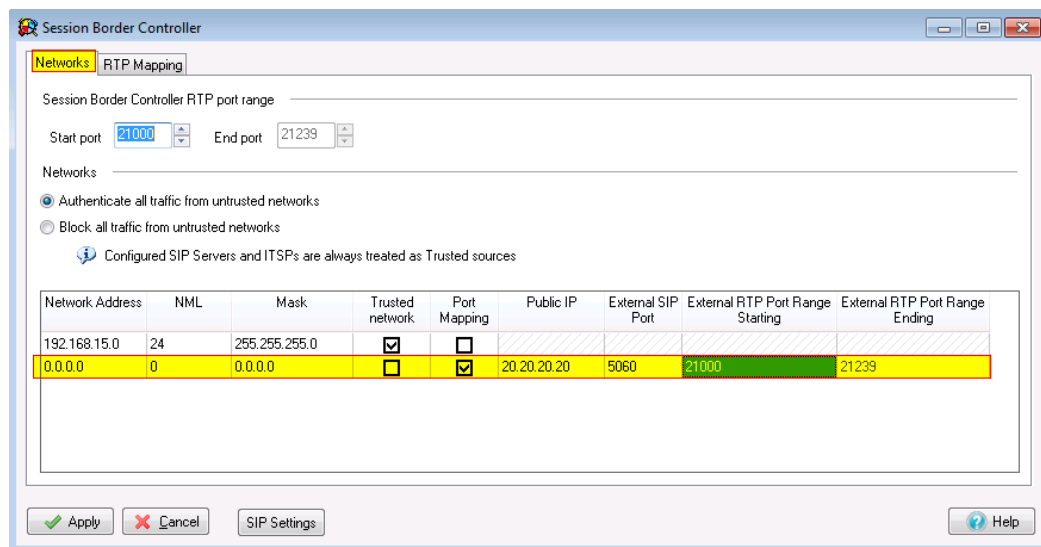
ALG was replaced / improved in MX version 5.0 by SBC. For full details on SBC please refer to the SBC document available from the KBS: <http://kbs.zultys.com/issue.php?bid=wp-998&fdwn=SBC.pdf> or from Technical Support document titled *sbc.pdf*, document number 0000000150.



Note: SBC only needs to be enabled if you are deploying your MX behind a firewall. If your MX is directly connected to the internet using the WAN or Ethernet 2 this section can be skipped.

In the example in the screenshot below:

- Network 192.168.15.0/24 is the private side (LAN) of the MX. This is where all the local phones are connected to the MX.
- Network 0.0.0.0/0 is mapping all other networks to the public side of the MX, thus performing the SBC functions to correct all RTP packets by inserting the public IP listed in the *Public IP* field, and the External SIP port with the port listed in the *External SIP Port* field.
- *Session Border Controller RTP Port Range* is the port range that the session border controller will use to send and receive voice packets.
- *External RTP Port Range* is the port range that is forwarded by the edge firewall to the MX, this port range is independent of the *Session Border Controller RTP Port Range* (in the example below, they were made to be the same).



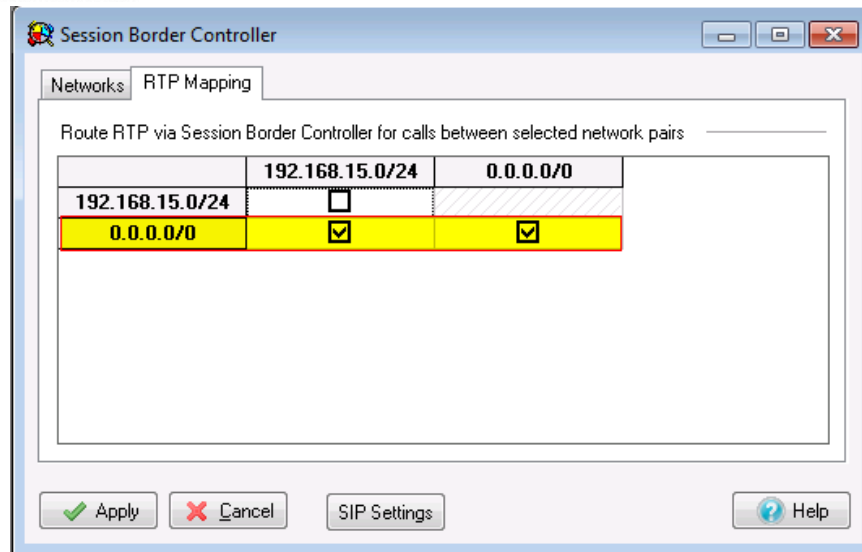
5.1 Networks

To configure SBC in the 0.0.0.0 route, enable port mapping, and assign the Public IP and ports in the fields circled in the screenshot above. If you are not using SBC and have the MX in routing mode with public IPs assigned to interface 2/WAN, do not check the Port Mapping box.

5.2 RTP Mapping

Check each network for the MX to perform ALG/SBC modifications for. RTP mapping must be done if using SBC or if your MX is directly connected to the internet.

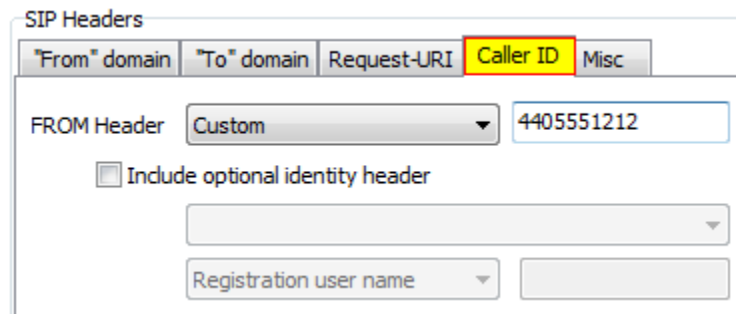
The 0.0.0.0/0 route must be checked for all networks as seen below.



6. Alternate Setup

6.1 Alternate Caller ID Setup

Caller ID can be set to *Custom* with a DID belonging to the customer.



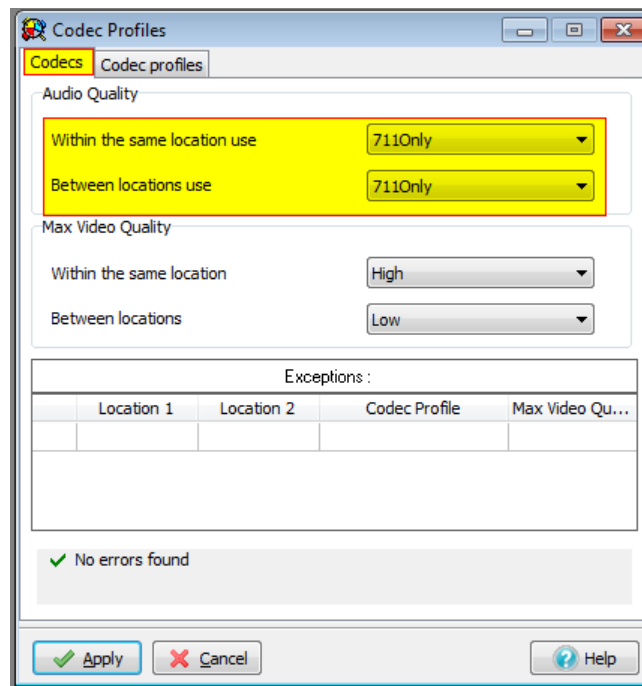
7. Requirements

7.1 General

- The MX must be running 9.0.4 or better to ensure proper functionality.
- P-Asserted Identity is not supported.
- Only G711 is supported by the carrier, a custom codec profile must be created.

7.2 MXnetwork

When the deployment includes a MXnetwork, you will need to modify the codecs used between nodes. This is accomplished by setting the codec profile applied to calls within the same location as well as between locations to the previously created custom codec profile.



The parameters that are set in the Codecs panel include:

- **Within the same location use:** This parameter determines the codec profile for voice calls between SIP devices at the same location.
- **Between locations use:** This parameter determines the codec profile for voice calls between SIP devices that are at different locations.
- **Exceptions:** This table determines the codec profiles for voice calls between SIP devices that are in two specific locations. Codec profile settings in this table override the default settings in the upper part of the panel.

To add an exception to the table, point the cursor in the table, right click your mouse, and select Add Exception. The Location 1 and Location 2 options are Location names, as configured on the Locations window.

Important: Codec panel changes do not take effect until you press the Apply button. If you press the Cancel button before pressing Apply, all pending changes to the Codec panel and the Codec Profile panel are disregarded. Pressing the Apply button saves all pending changes all Codec Profile panels.