



Zultys PBX
SIP Trunking
Setup & User Guide

Nextiva.com
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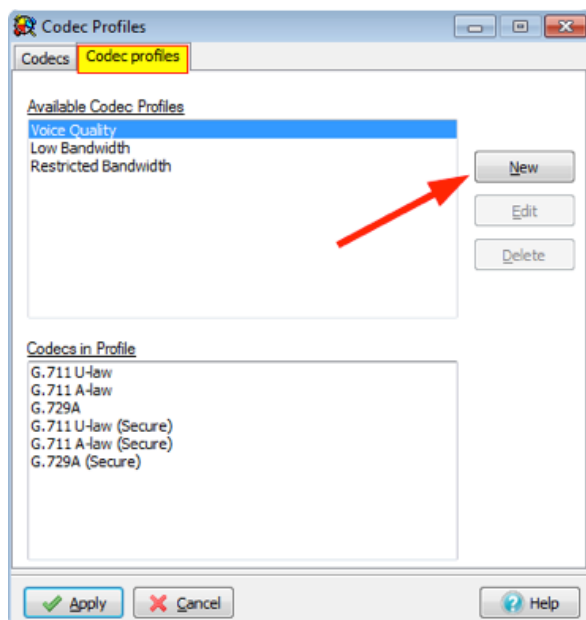
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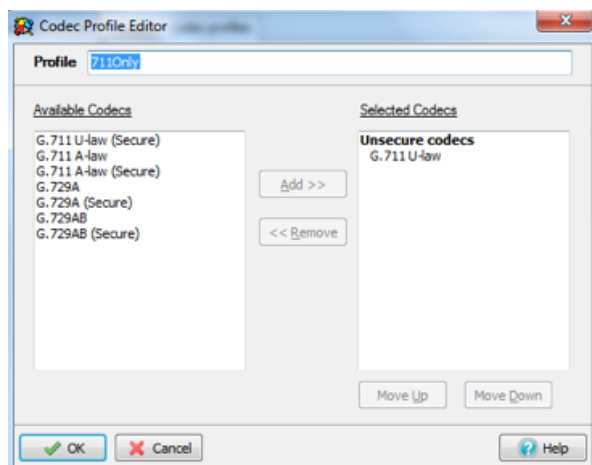
NOTE: In this document all IP addresses are example IP addresses, please replace with a valid IP of your MX

Create Custom Codec Profile

Nextiva supports G711 and G729, but under certain conditions will not properly negotiate the correct codec. When this condition occurs, it is recommended to set 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 creates and edits Codec Profiles. You access the Codec Profile Editor by pressing the **New** or **Edit** button in the Codec Profiles window.

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.

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 will be 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.

Modifying Profile Parameters

To add codecs to the current profile, select one or more codecs in the Available Codecs table 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 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.

Button Bar

OK: Pressing the OK button closes the Codec Profile Editor and places the changes in the Codec Profiles window. These changes are only permanently saved to the database 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.

Create Custom SIP Profile

Create a new profile named “**No Re-INVITE**”, select the option not to send Re-INVITEs and to look for the DID numbers in the “To”. Click the **OK** button to save the options and close the window.

To create a new SIP Profile, click on the “**SIP Profiles**” button and then right-click and choose **New**.

Profile Table

The profile table lists the defined SIP profiles available to SIP Servers. Profiles listed in bold text with a padlock are provided by Zultys. You cannot edit or remove these provided profiles.

- To create a new profile, right-click in the profile list and select **New**.
- To copy an existing profile as a new profile, right-click in the profile list and select **Duplicate**.
- To edit an existing profile, highlight the profile and select the preferred parameter options.
- To remove an existing profile, right-click on the desired profile to highlight it, then select **Delete** in the popup menu.

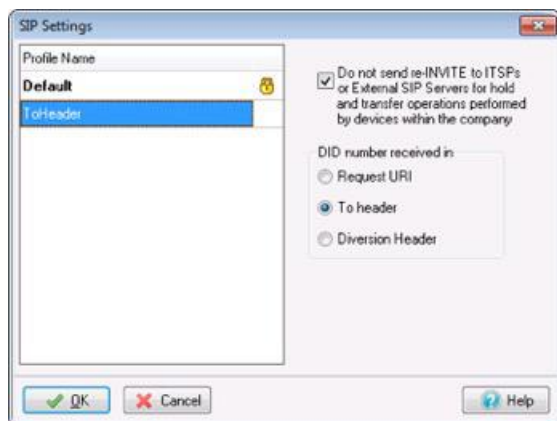
Parameter Table

Parameter table settings are applied to each circuit to which the profile is assigned.

- Do not send Re-INVITE for internal hold and transfer operations: This parameter affects the MX operation when calls involving external devices are placed on hold or transferred.
- When this option is selected, the MX responds to a hold or transfer request by the internal phone by playing music on hold for the external device. When the call is resumed, the MX replaces the music on hold with the new audio stream from the internal device.
- When this option is not selected, the MX responds to a hold or transfer request by the internal phone by sending a Re-INVITE to the external device, which causes the recipient to stop sending media packets. The call is resumed through the sending of a subsequent Re-INVITE.
- DID number received in: This parameter specifies the line within a SIP request that lists the DID number of the recipient.

Press the **OK** button to accept all SIP Settings changes and return to the SIP Servers and ITSP panel. Changes to the SIP Settings panel are not saved to the database until the **Apply** button on the SIP Servers and ITSP panel is subsequently pressed.

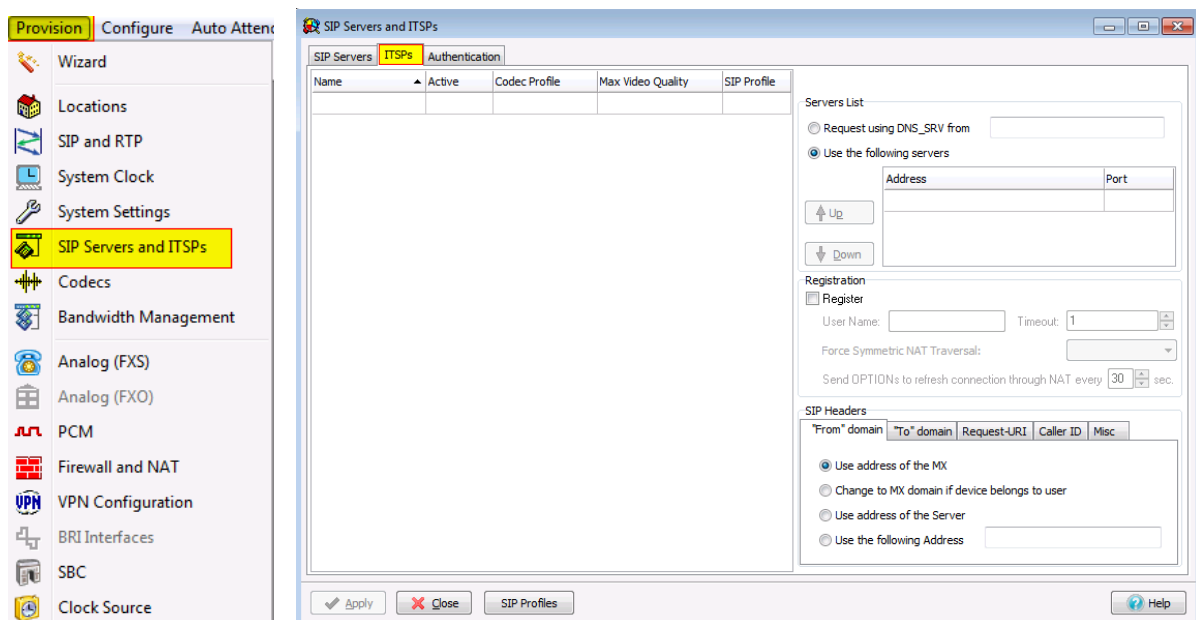
Press the **Cancel** button to discard all SIP Settings changes and return to the SIP Servers and ITSP panel. Changes to the SIP Settings panel can also be discarded by pressing the **Cancel** button on the SIP Servers and ITSP panel.



Right-click to create a new profile, name it *To header*. Select the option that instructs the MX to look for the DID number in the *To header*.

Create ITSP Account

In the MX Administrator, go to *Provision* and 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, and 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.

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.
- **SIP Transport:** This parameter specifies which SIP signalling transport is used. Options include UDP, TCP or TLS. Default value is UDP.
- **Type:** This parameter specifies how the MX handles incoming calls from the SIP server. Valid parameter settings include:
 - **Internal:** The number specified in the SIP INVITE is treated as a dialling 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 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, and select **Delete** from the menu.

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.
- **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 behaviour 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 port parameter, if it is behind the NAT, and if so, enforces NAT binding using the 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 the 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.

Configure the Account

Enter the following information:

- **Name:** Nextiva
- **Active:** Check
- **SIP Transport:** UDP (only available in MX version 10 or better)
- **Codec Profile:** Select *Voice Quality OR 711Only (Custom profile created earlier)*
- **SIP Profile:** Select *To Header (Custom profile created earlier)*
- **Video Codec:** Select *None*
- **Use the following servers:** *bt.voipdnsservers.com*
- **Port:** 5060
- **Register:** Checked
- **Username:** The User ID provided by Nextiva
- **Force Symmetric NAT Transversal:** Off
- **From header:** Select *Use the following Address: bt.voipdnsservers.com*
- **To header:** Select *Use custom domain name: bt.voipdnsservers.com*
- **Request-URI:** Select *FQDN*
- **Caller ID:** Select Custom with the User ID provided by Nextiva
 - Include Optional identity header
 - P-Asserted-Identity
 - Originating party Caller ID
- **Misc:** Nothing selected

The screenshot displays the 'SIP Servers' configuration page in a web interface. The 'Authentication' tab is active. A table lists the configured servers:

Name	Active	SIP Transport	Codec Profile	Max Video Quality	SIP Profile
Nextiva	<input checked="" type="checkbox"/>	UDP	Voice Quality	None	Toheader

The configuration panel for the 'Nextiva' server is shown on the right. It includes the following settings:

- Servers List:**
 - Request using DNS_SRV from
 - Use the following servers

Address	Port
bt.voipdnsservers.com	5060
- Registration:**
 - Register
 - User Name: NextOS User ID
 - Timeout: 3600
 - Force Symmetric NAT Traversal: Off
 - Send OPTIONs to refresh connection through NAT every 30 sec.
- SIP Headers:**

"From" domain	"To" domain	Request-URI	Caller ID	Misc.
<input type="radio"/> Use the domain name for the appropriate server in the Servers List				
<input type="radio"/> Use resolved IP address for the appropriate server in the Servers List				
<input checked="" type="radio"/> Use custom domain name	bt.voipdnsservers.com			

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

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

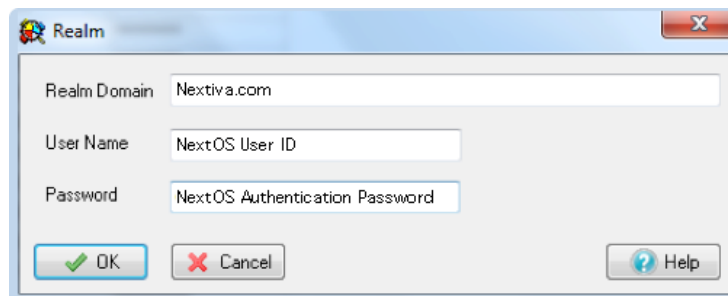
SIP Headers

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

Ignore Domain received in From/To SIP headers
 Unsupported media type response code

Authentication

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



- Realm Domain: nextiva.com
- User Name: authentication name provided by Nextiva
- Password: password provided by Nextiva

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.

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, and select **Edit**.

To remove a Realm from the table, select the Realm, right-click, and select **Delete** from the menu.

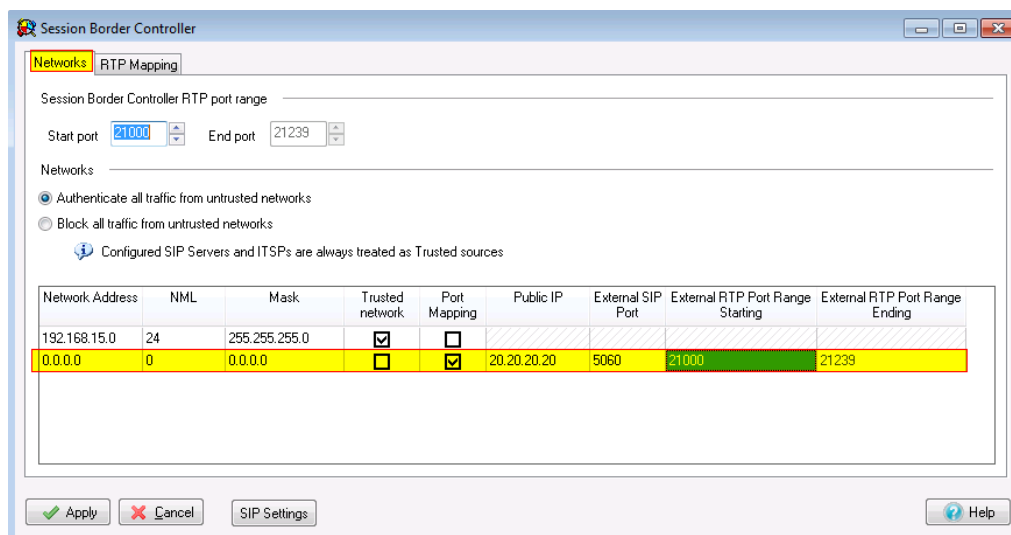
Enabling ALG / SBC on MX (MX version 5.0+)

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 the 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).



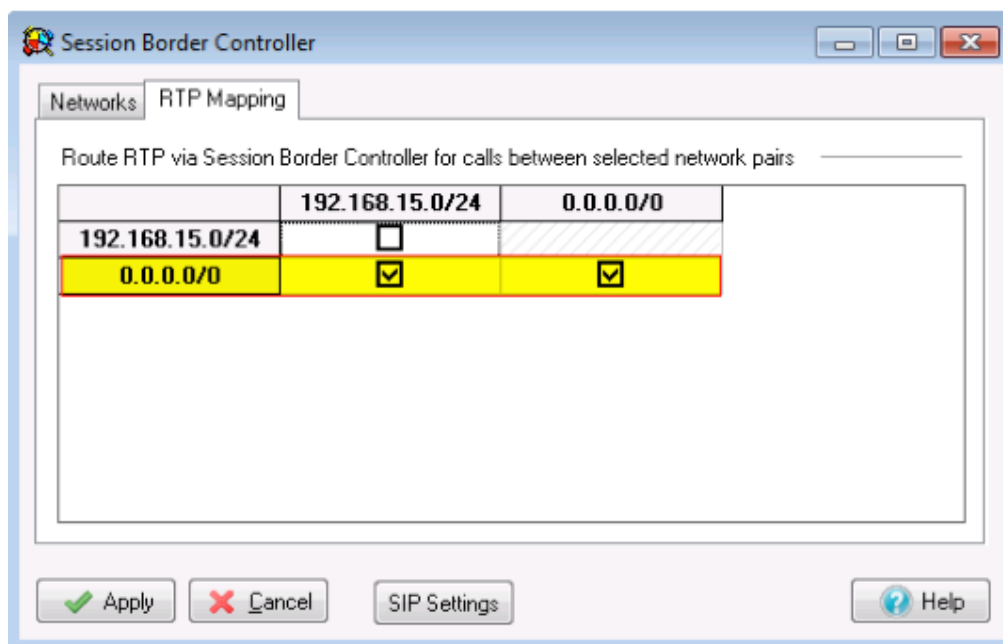
Networks

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.

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.



Requirements

General

- The MX must be running 9.0.4 or better to ensure proper functionality.
- In some cases when using Voice quality, codec re-negotiation might fail. If this is a problem, set G711only or G729only.

CID

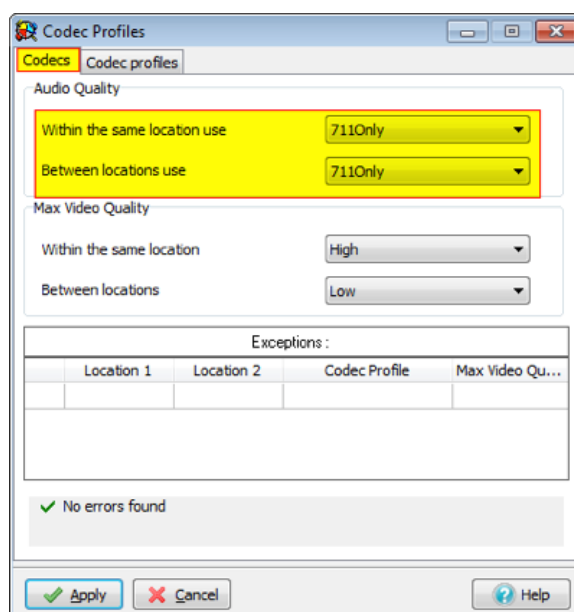
Incoming calls: if the caller has CID <> 10 digits, the call comes in as anonymous.

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, 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 for all Codec Profile panels.