



Accessline ITSP Setup

Author: Zultys Technical Support

This configuration guide was created to assist knowledgeable vendors with configuring the Zultys MX Phone System with Accessline service.

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

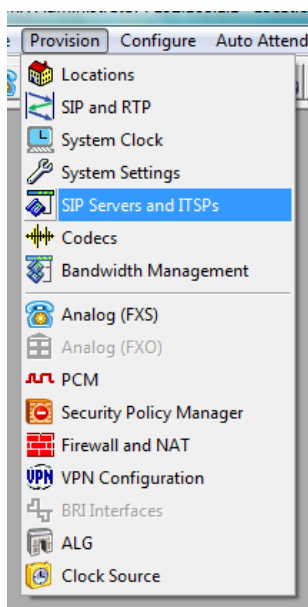
For administrative or technical questions related to Accessline contact Accessline directly.

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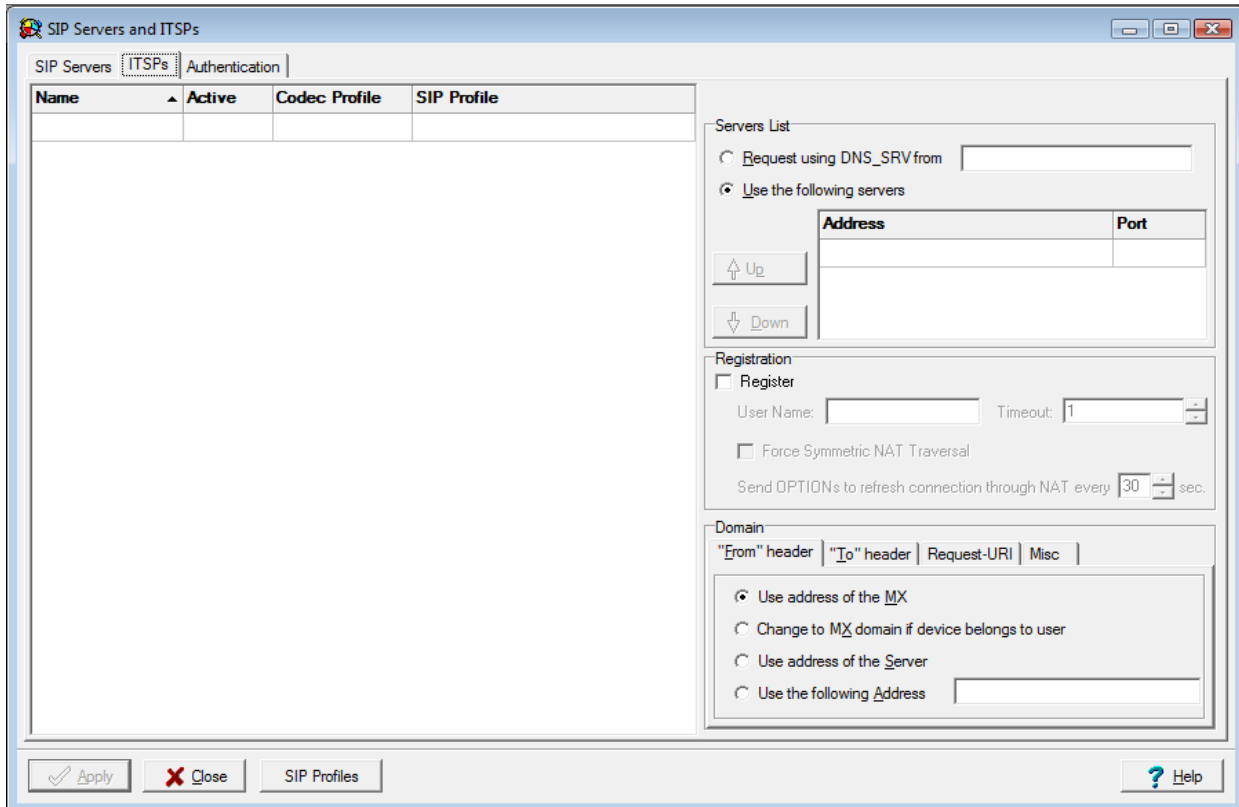
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1.1 Create ITSP Account

In the Admin UI goto Provision → SIP servers and ITSPs



This will bring up the SIP Servers and ITSP configuration screen. It has three (3) tabs along the top. To configure SIP server click on the *SIP* tab, for ITSPs click on the *ITSP* tab, and to set up the authentications click on the last tab *Authentication*.



1.1.1 The SIP Servers and ITSPs panels each comprise two sections:

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

1.1.2 Servers Table

The Servers table in the ITSPs panels lists the ITSP 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 ITSP server.

- **Name:** This parameter identifies the SIP Server to the MX. Other UI windows, such as the Dial Plan: Routing panel, references 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 the method that 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. Press 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 in 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.

1.1.3 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 desired 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, place the cursor in 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:** Place a mark in this selection box to enable the MX to register as a client to the specified SIP Server.
 - **User Name:** This parameter specifies the string that is specified as the user name in the From field for INVITE packets sent from the MX to the SIP Server if the Registration parameter is enabled. The From field derives the Domain name on the basis of the Domain in From Header parameter.
 - **Timeout:** This parameter specifies the registration period for the MX. This parameter is valid only if the Registration option is selected.
 - **Force Symmetric NAT Transversal:** This option regulates the way how RFC3581 "An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing" is used. There are 3 possible values for this option:
 - Off - RFC3581 defined behavior is disabled.
 - On - MX always enforces RFC3581 when communicating to this ITSP. OPTIONS SIP message is used to maintain NAT binding.
 - Auto - MX tries to learn via rport parameter if it's 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's not recommended to use this configuration option anymore.

- **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 specify the MX as the originator address.
 - Select Change to MX domain if device belongs to user to specify the MX as the originator address if the MX receives the message from an MX User. If the message is received from an unknown user (such 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.

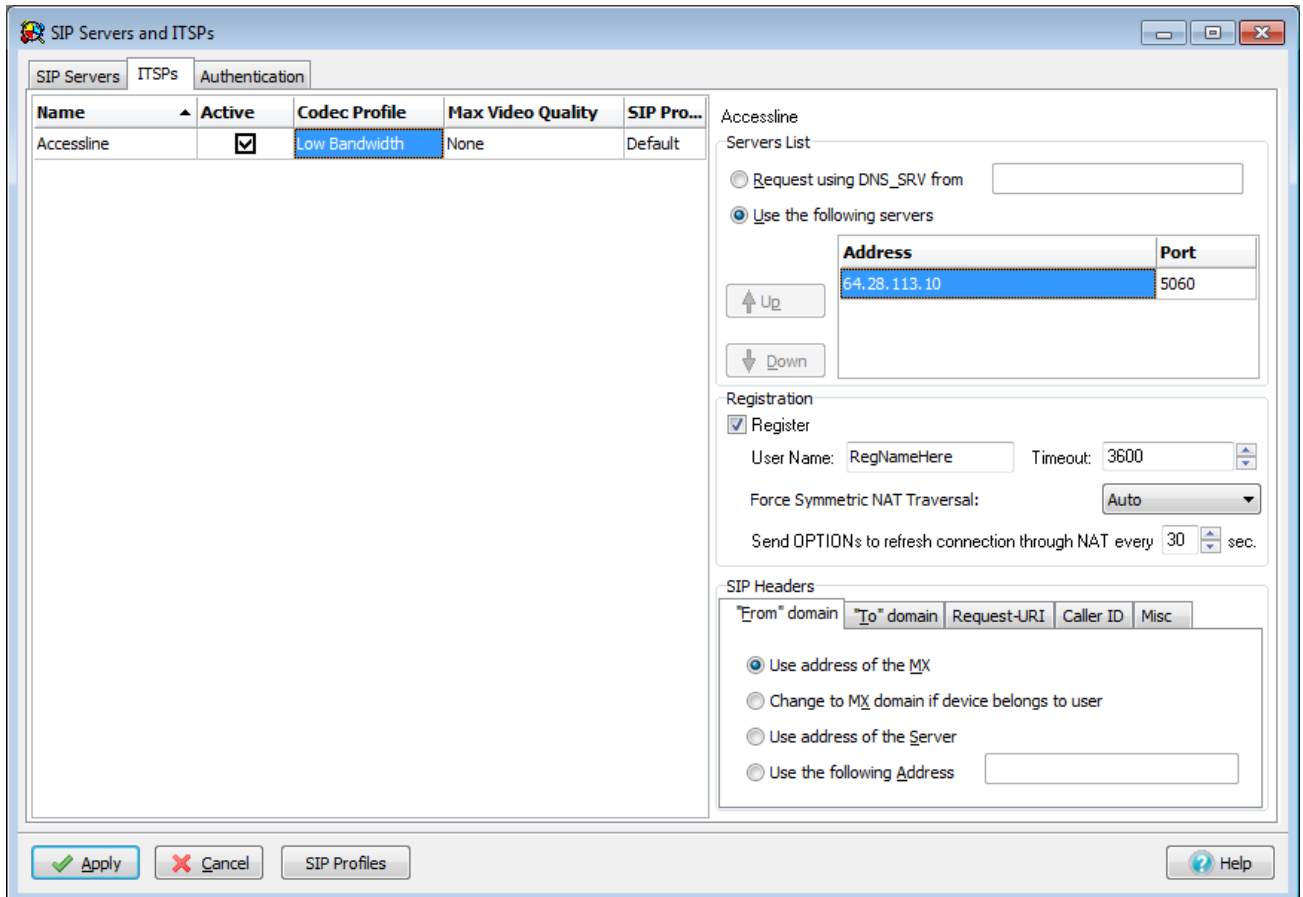
1.2 Configure the account

The following section(s) contain URLs and Domain Names that should only be viewed as examples. These details are Accessline account and region specific. You must use the detailed information that is provided to you on the Accessline 'Install profile' provided by your Accessline Service Coordinator

Enter the following information

- **Name:** Accessline
- **Active:** Check
- **Codec Profile:** Low bandwidth
- **SIP Profile:** Default
- **Use the following servers:** the IP address given to you by Accessline (See the Accessline "Install profile").
- **Port:** 5060
- **Register:** Check
- **Username:** user name/phone number given by Accessline (See the Accessline "Install profile").

- **From domain:** Use the address of the MX.
- **To domain:** Use resolved IP address of the appropriate server in the Servers List.
- **Request-URI:** IP Address
- **Caller ID:** Caller ID of the originating user/call group
- **Misc:** Nothing checked



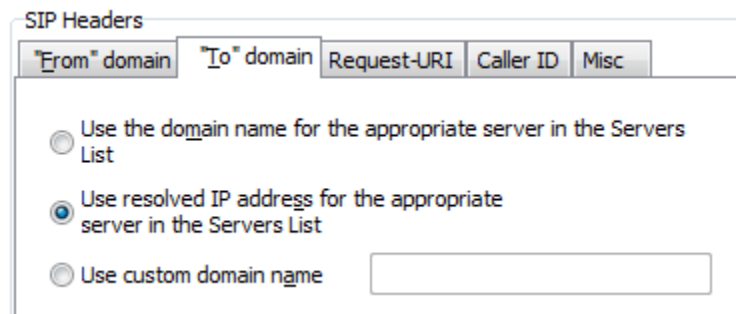
The screenshot shows the 'SIP Servers and ITSPs' configuration window. The 'SIP Servers' tab is active, displaying a table with the following data:

Name	Active	Codec Profile	Max Video Quality	SIP Pro...
Accessline	<input checked="" type="checkbox"/>	Low Bandwidth	None	Default

The 'Accessline' configuration panel on the right includes the following sections:

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

Address	Port
64.28.113.10	5060
- Registration:**
 - Register
 - User Name: RegNameHere 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 []



The close-up shows the 'SIP Headers' configuration with the '*To* domain' tab selected. The options are:

- 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

Based on service type
 Caller ID of originating user/call group
 Custom

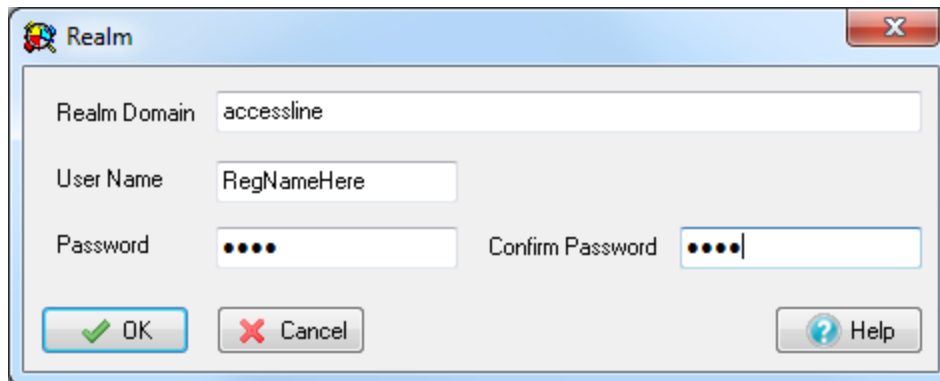
SIP Headers

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

Ignore Domain received in From/To SIP headers

1.3 Authentication

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



The realm for Accessline is “accessline” (unless otherwise indicated)

- **Realm Domain:** accessline
- **User Name:** User name is the username provided by Accessline
- **Password:** Password provided by Accessline
- **Confirm Password:** Password provided by Accessline

1.3.1 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. This 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.

1.3.2 Editing the Table

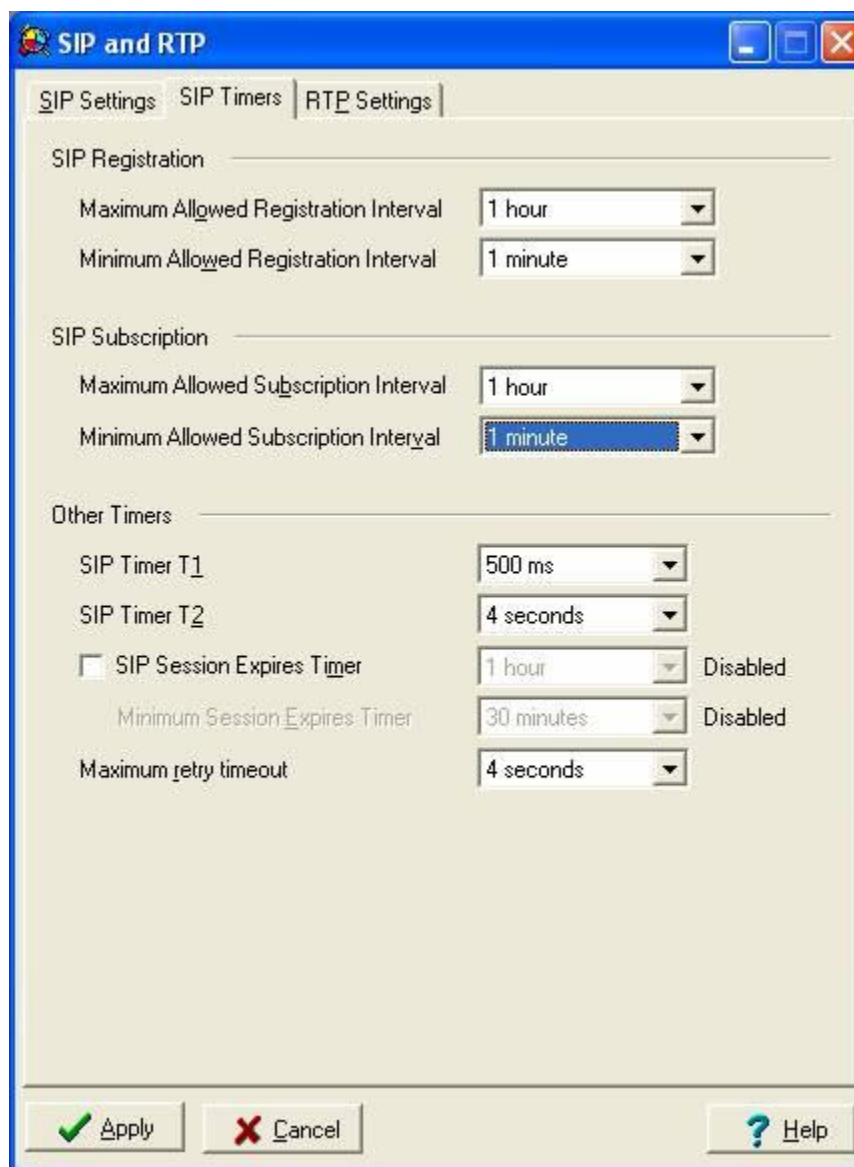
To add to a realm, access the Realm panel by right clicking in the table and selecting Add.

To edit a realm, access the Realm panel by double clicking in one of the Realm entries cells or select a Realm, press the right mouse button, and select Edit.

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

1.4 Disable Session Timers

Clear the check mark from *SIP Session Expires Timer*. This will require a system reboot. This can be found under *Provision | SIP and RTP*.



The screenshot shows the 'SIP and RTP' configuration window with the 'SIP Timers' tab selected. The window contains the following settings:

Section	Parameter	Value	Status
SIP Registration	Maximum Allowed Registration Interval	1 hour	
	Minimum Allowed Registration Interval	1 minute	
SIP Subscription	Maximum Allowed Subscription Interval	1 hour	
	Minimum Allowed Subscription Interval	1 minute	
Other Timers	SIP Timer T1	500 ms	
	SIP Timer T2	4 seconds	
	<input type="checkbox"/> SIP Session Expires Timer	1 hour	Disabled
	Minimum Session Expires Timer	30 minutes	Disabled
	Maximum retry timeout	4 seconds	

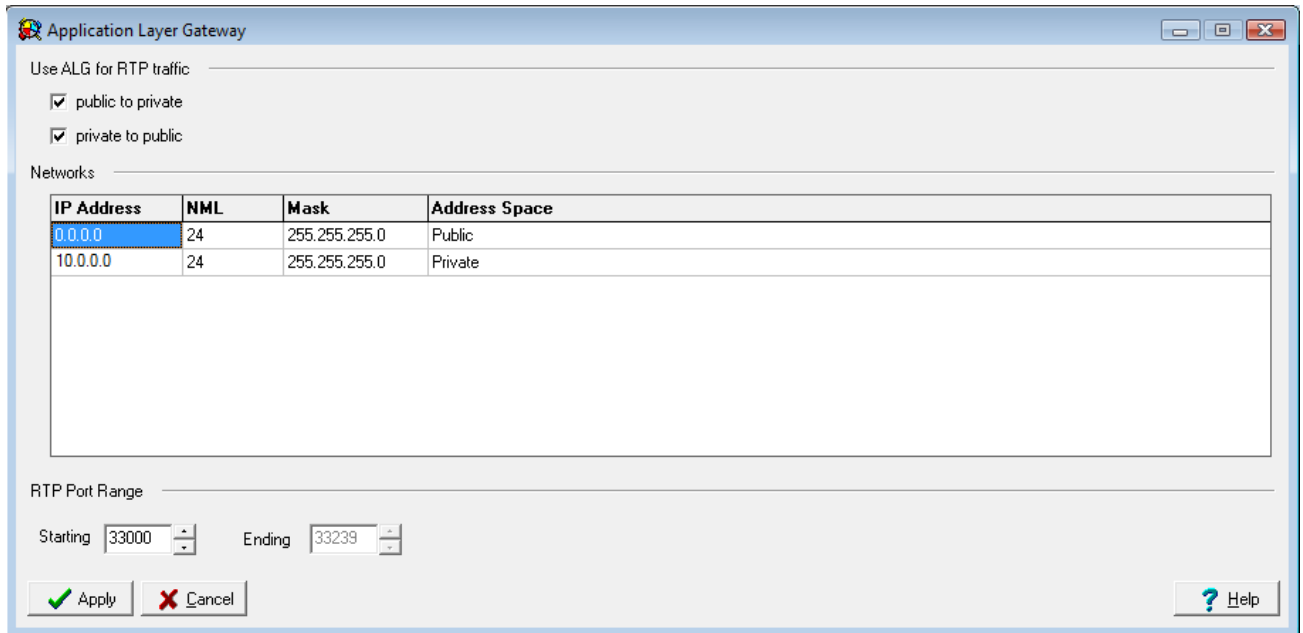
At the bottom of the window are three buttons: 'Apply' (with a green checkmark icon), 'Cancel' (with a red X icon), and 'Help' (with a blue question mark icon).

1.5 Set Up ALG (MX version 3.2.10 to 4.2.14)

Set up ALG in *Provision* | *ALG*

For the example below:

- Network 10.0.0.0/24 is the private side (LAN) of the MX, this is where all the local phones are connected to the MX.
- Network 0.0.0.0/24 is mapping all other network to the public side of the MX, thus performing the ALG functions to correct all RTP packets.



Application Layer Gateway

Use ALG for RTP traffic

public to private

private to public

Networks

IP Address	NML	Mask	Address Space
0.0.0.0	24	255.255.255.0	Public
10.0.0.0	24	255.255.255.0	Private

RTP Port Range

Starting Ending

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

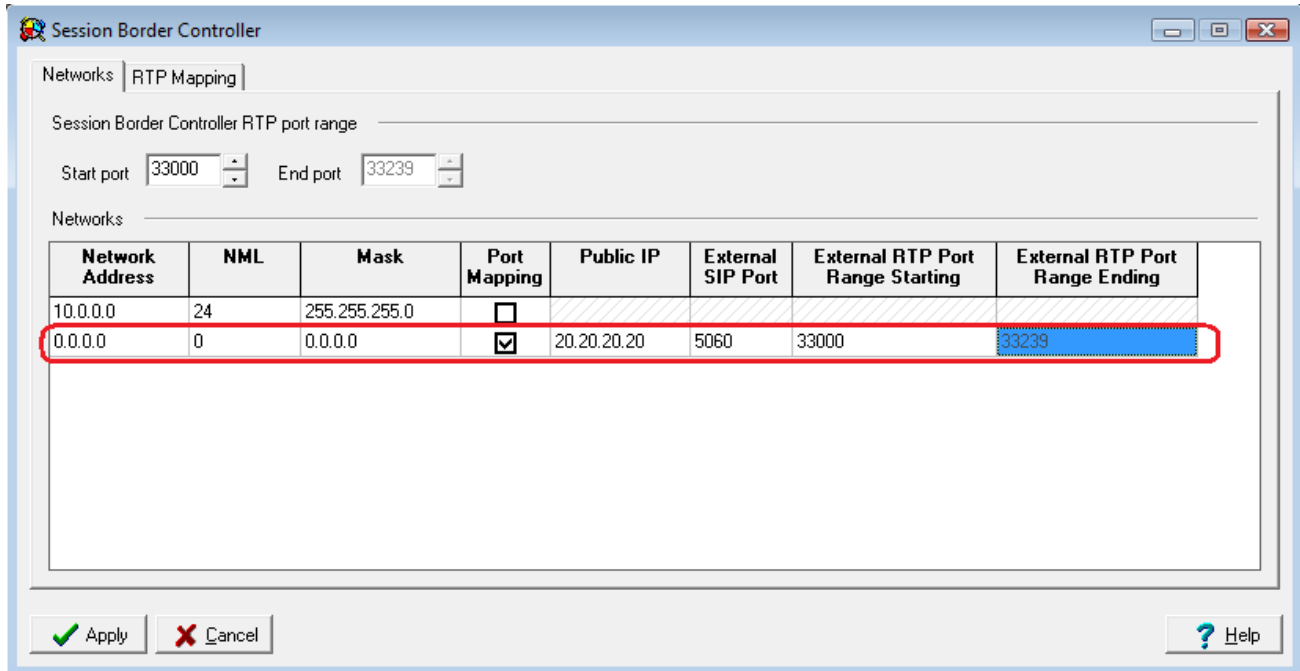
ALG was replaced / improved in MX version 5.0 by SBC. For full details on SBC please refer to the SBC document available from the KBS <http://kbs.zultys.com/issue.php?bid=wp-998&fdwn=SBC.pdf> or from Technical Support named *sbc.pdf* document number 0000000150.

For the example below:

- Network 10.0.0.0/24 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 network 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* are the ports that the session border controller will 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 there were made to be the same.)

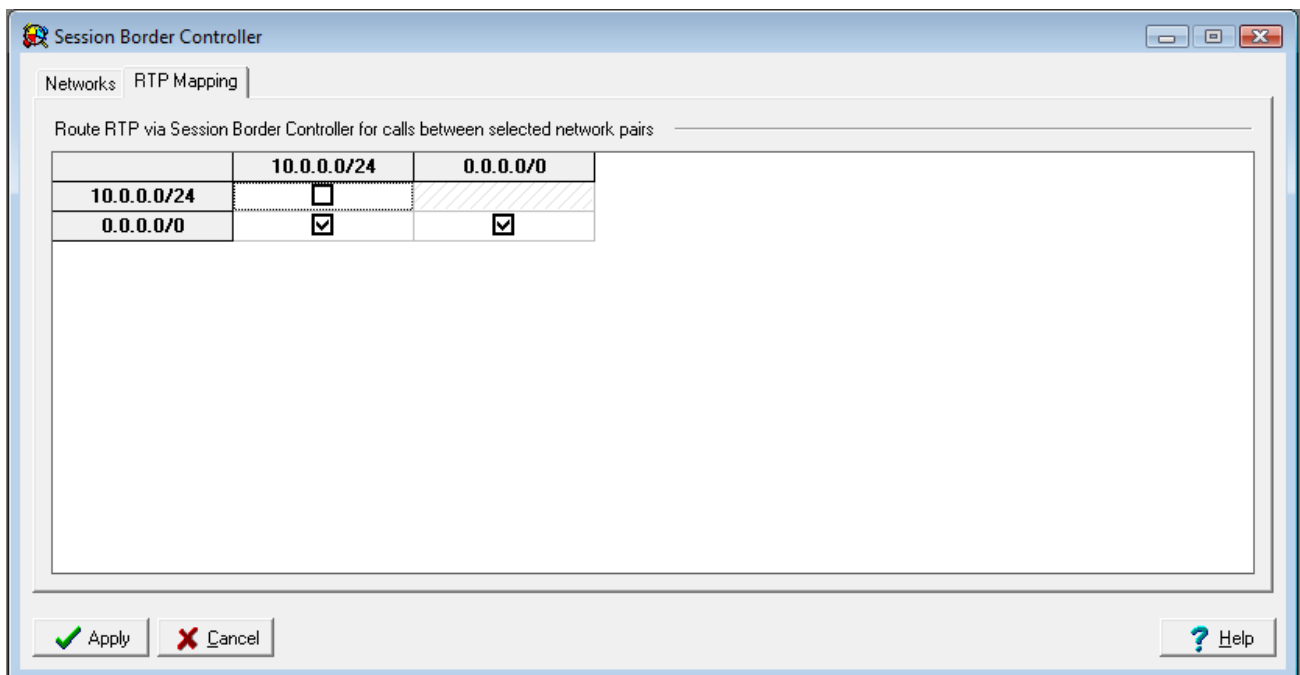
1.6.1 Networks

To configure SBC in the 0.0.0.0 route enable port mapping, and assign the Public IP and ports in the fields circled below. 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.



1.6.2 RTP Mapping

Place check each network you wish the MX to perform ALG/SBC modifications for.



Note 1: Sip trunk licenses are required by MX in order to place/receive calls via ITSP, MX firmware's **PRIOR** to version 5.0. Starting with Version 5.0 SIP Trunk licenses are no longer needed.