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Broadvox Fusion Platform Version 1.2
ITSP Setup Guide

Author: Zultys Technical Support

This configuration guide was created to assist knowledgeable vendors with configuring the Zultys MX Phone System with Broadvox Fusion platform version 1.2 ITSP service.

The Zultys MX must be running Release 8.0.7 or later for proper operation, in order to receive support as a certified deployment from Broadvox.

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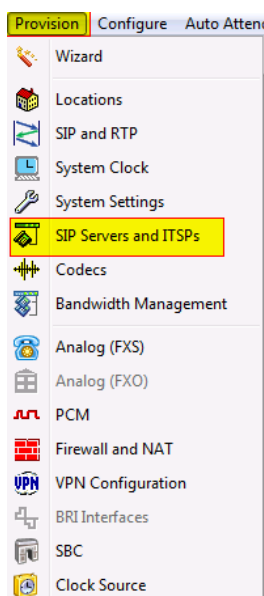
NOTE: in this document all IP addresses are example IP addresses, please replace with a valid IP of your MX

Content

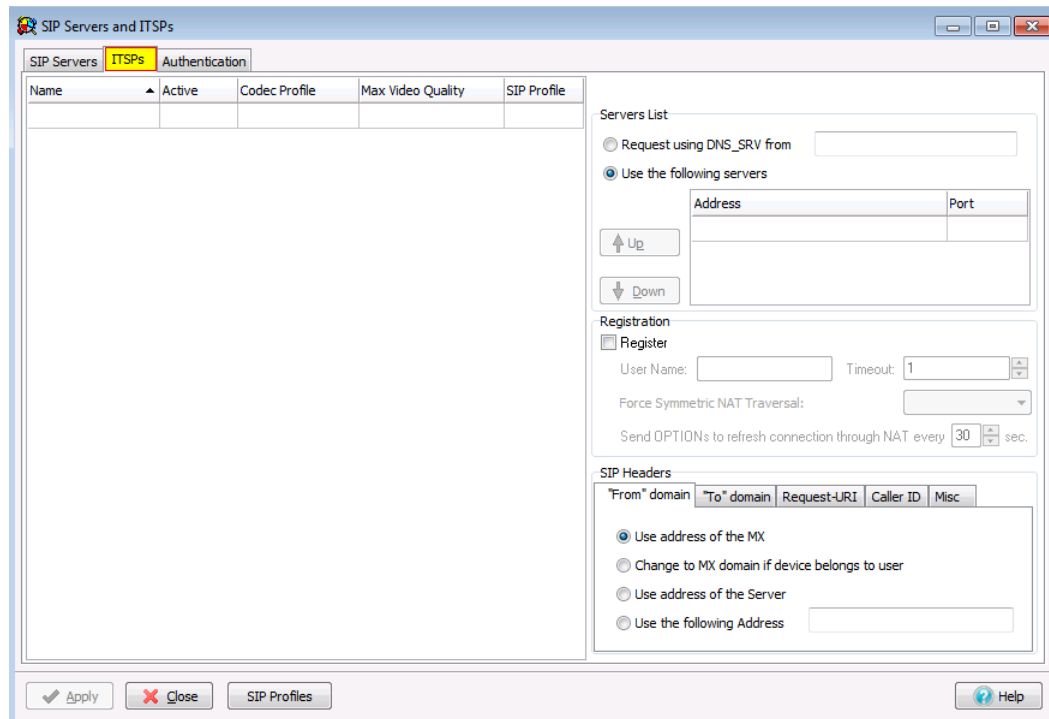### 1.   Create ITSP Account

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.

1.1 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:

- o **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.

- o **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 new information.

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

1.2   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:

- o **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.

- o **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.

- o **Registration**: This section specifies the registration parameters that allow the MX to register as a client to the selected SIP server.

- o **Register**: Check this box to enable the MX to register as a client to the specified SIP Server.

- o **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.

- o **Timeout**: If the Registration option is enabled, this parameter specifies the registration period for the MX.

- o **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.

- o **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.

o **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.

o **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.

o **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
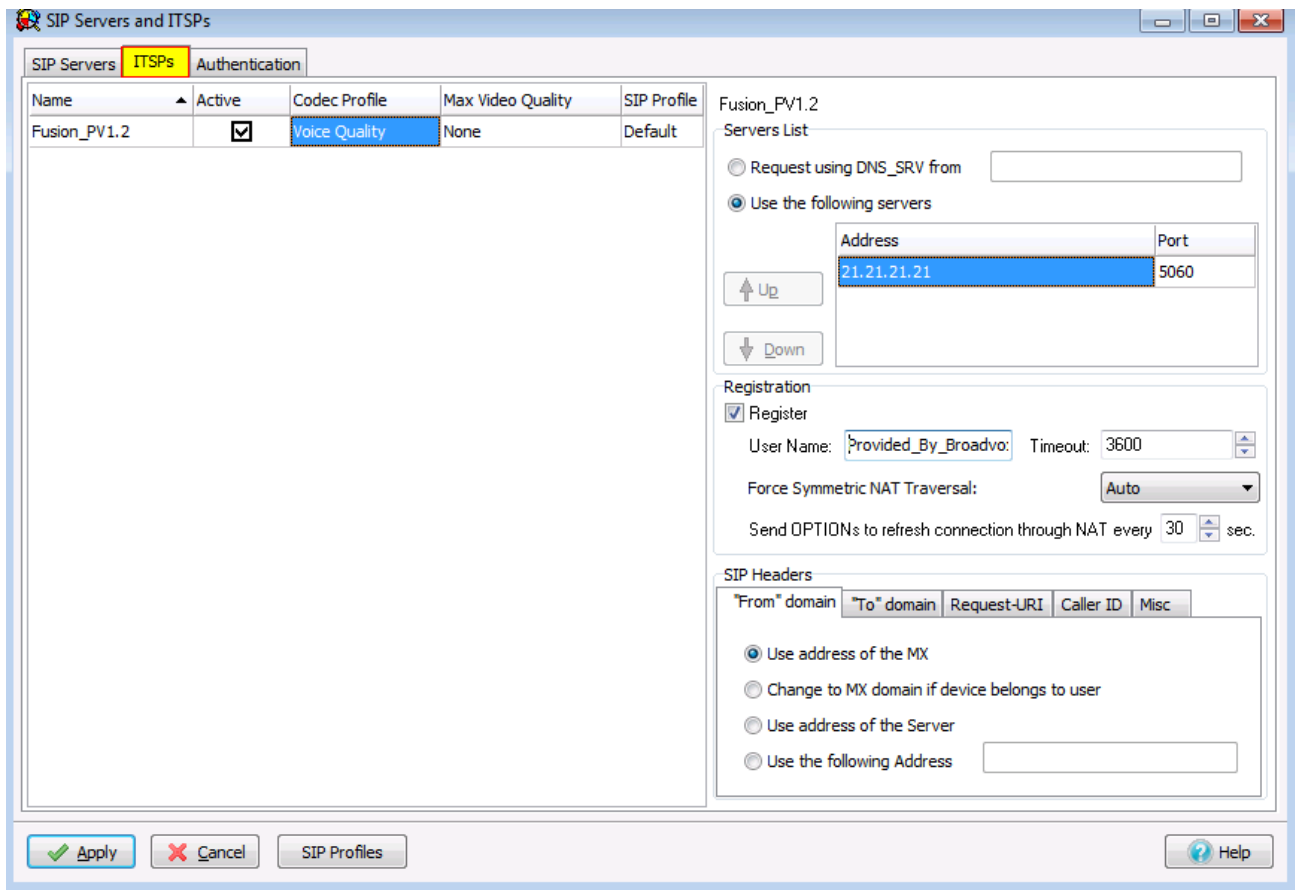
- Include optional identity header: Select

- P-Asserted-Identity: to populate the P-Asserted Identity field with the following property

    o Registration User Name: Populate the Caller ID as the Registration User Name

    o Originating party Caller ID: Populate Caller ID as the Originating Party Caller ID

    o Custom: Populate Caller ID with the custom value entered

- Remote Party ID: to populate the Remote Party ID field with the following property

    o Registration User Name: Populate the Caller ID as the Registration User Name

    o Originating party Caller ID: Populate Caller ID as the Originating Party Caller ID

    o Custom: Populate Caller ID with the custom value entered

o **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

2. **Configure the account**

Enter the following information:

- **Name**: Fusion PV1.2
- **Active**: Check
- **Codec Profile**: Select *Voice Quality* or *Low Bandwidth*
- **SIP Profile**: Select *Default*
- **Video Codec**: Select *None*
- **Use the following servers**: *Provided by Broadvox*
- **Port**: 5060
- **Register**: Checked
- **Username**: Provided by Broadvox
- **Force Symmetric NAT Transversal**: Auto
- **From header**: Select *Use Address of the MX*

- **To header**: Select *Use resolved IP for the Appropriate Server in the Servers List*
- **Request-URI**: Select *IP Address*
- **Caller ID**: Select *From header*, and enter Registration user name
- **Misc**: Nothing selected

3. Authentication

The *Authentication* tab is used to define the authentication information for realm based authentication.
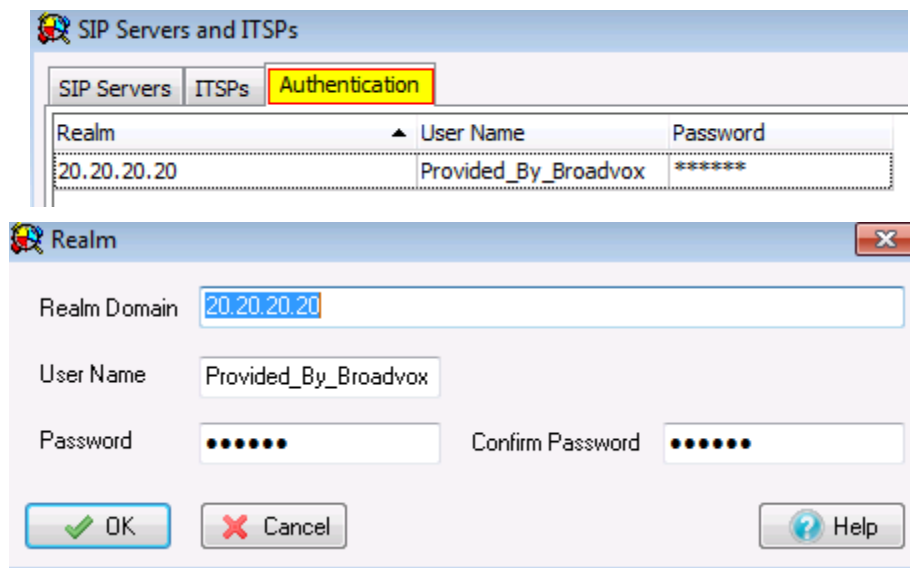
The realm for Broadvox Fusion Platform V1.2 is the public IP of the MX itself

- **Realm Domain:** Public IP of the MX itself.
- **User Name:** User name is the username provided by Broadvox.
- **Password:** Password provided by Broadvox.
- **Confirm Password:** Password provided by Broadvox.

4. 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).
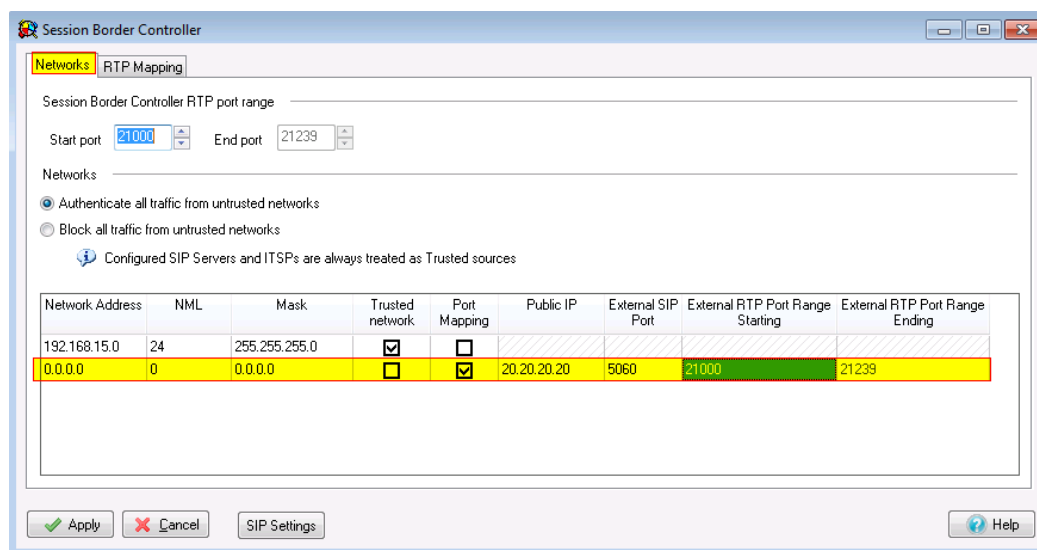

4.1 Networks

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.

4.2 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.
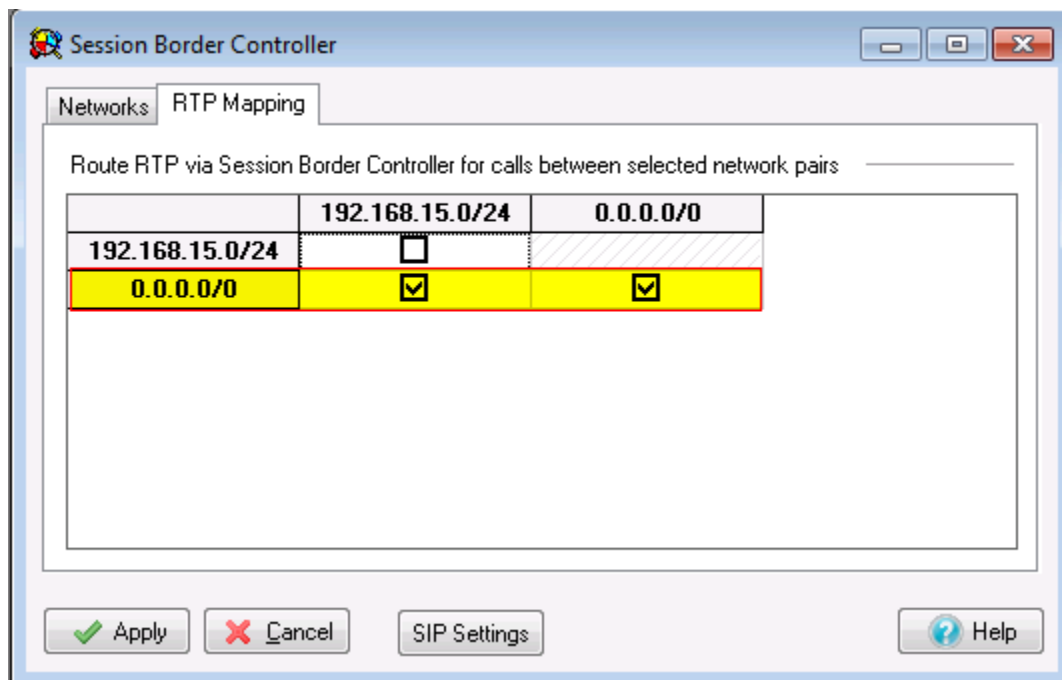
5. Requirements

- The MX must be running MX Release 8.0.7 or later to ensure proper functionality.
- Fax messaging may not be 100% reliable for some numbers.
- T38 cannot be disabled by Broadvox.