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MAXCS Release 7.0

Application Note: Remote Survivability with EdgeMarc SBC

Intended audience:
AltiGen Authorized Partners

May 7, 2014

Contents

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About This Guide

With a standard MaxCS installation, an internet outage can prevent the on-premise IP phones or remote branch phones from reaching the MaxCS 7.0 server.

This guide explains how to configure local PSTN survivability in an on-premise site or a remote branch office, so that when an Internet outage occurs, IP phone users can still make outbound calls through a local PSTN trunk via an EdgeMarc Session Border Controller (SBC).

The examples in this guide are based upon a configuration of an AudioCodes MP-118 with 4 FXS and 4 FXO ports.

This guide comes with two files. (These two files can be found in the AltiGen Knowledge Base, along with this guide.) Store these files locally; you will need to upload them into the AudioCodes configuration tool during the procedures in this guide (if you are using the EdgeMarc SBC with an AudioCodes MP118 gateway).

- usa_tones_13_NoHold.ini
- usa_tones_13_NoHold.dat

IMPORTANT: AltiGen does not provide general configuration support for EdgeWater or AudioCodes products.

A support agreement with EdgeWater or AudioCodes is required.

Related Documents

- For instructions on configuring Polycom IP phones, refer to the *Polycom IP Phone Configuration Guide.*
- For instructions on configuring analog extensions using an AudioCodes MP1xx gateway, refer to *Configuring Analog Extensions with AudioCodes Gateways*. This guide is available from your AltiGen representative.

Introduction

When an Internet outage prevents the on-premise IP phones from reaching the MaxCS server, IP phone users can still make outbound PSTN calls through a local PSTN trunk connected to the FXO port of the MP1xx gateway. Inbound PSTN calls to the local PSTN trunk can still be routed to a local extension.

If you already have an AudioCodes MP1xx FXS/FXO gateway with analog phones that log in to MaxCS as IP extensions, you can reuse the same MP1xx gateway to configure the survival FXO ports connecting to local PSTN trunks.

You also can have another MP1xx gateway to configure the survival FXO ports if you do not want to, or cannot, reuse the MP1xx box. The configuration is very similar.

Firewall Considerations

We recommended that you replace your original firewall/NAT with the EdgeMarc SBC in on-premise deployments or remote branch offices.

If you prefer to keep your original firewall/NAT, we recommend that you consult with Edgewater's Technical Support staff for more information.

Architecture

Figure 1 depicts a survival configuration with an EdgeMarc SBC and an AudioCodes MP118 gateway.

While the EdgeMarc SBC (10.40.0.95) can reach MaxCS (10.40.1.43), the phone users of extension 162, 161 and 169 can make calls, receive calls, and use the full features in MaxCS.

In the figure, PSTN user (5102520004) dials 510-252-0000. The AudioCodes MP118 gateway receives the call, and then sends it to MaxCS (10.40.1.43). The call is treated as a SIP Trunk call in MaxCS.

Note: An FXO door phone operates in the same manner as an incoming call.

Each organization can choose whether to utilize the SIP trunk to make a PSTN outbound call through the AudioCodes MP118 by configuring outcall routing in MaxCS MaxAdministrator. This can be done using Enterprise Manager or via a SIP TRUNK configured on the MaxCS pointed to the EdgeMarc device.

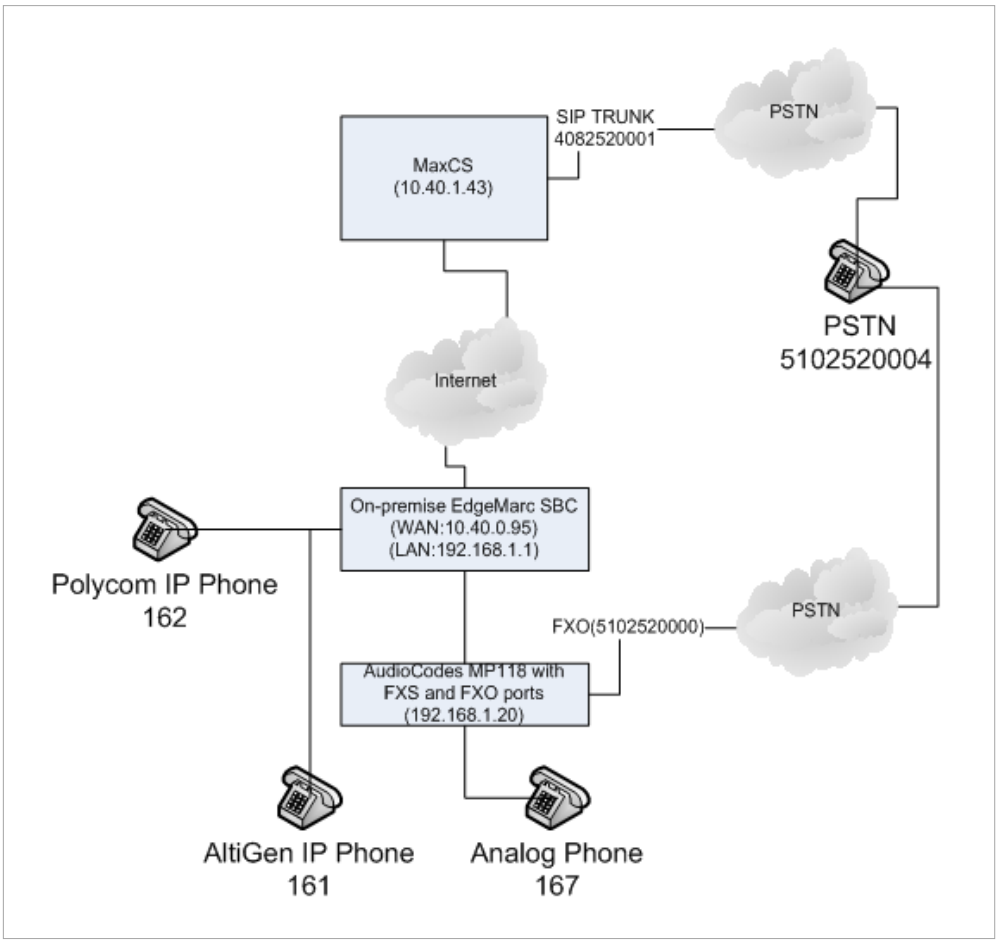


Figure 1: Remote survivability configuration

If the EdgeMarc SBC cannot reach MaxCS, PSTN outbound calls from extension 162, 161, and 169 will be routed to the AudioCodes MP118's FXO port by the EdgeMarc SBC automatically. In addition, extensions 162, 161 and 169 still can call each other. However, users will lose many of the features provided by MaxCS, such as voicemail, transfer, conference, and so on.

When the EdgeMarc SBC cannot reach MaxCS, incoming calls to 510-252-0000 will be routed to a pre-configured extension, (extension 161 in this example), through the EdgeMarc SBC.

Once the EdgeMarc SBC can reach MaxCS again, users will regain the full features of MaxCS within 2 minutes.

Prerequisites

In order to implement remote survivability via an EdgeMarc SBC, your environment must meet the following minimum requirements.

- Your must be running either MaxCS Release 7.0 Enterprise or MaxCS Private Cloud.
- You must have an EdgeMarc Session Border Controller (SBC).
- You must have an AudioCodes MP-1xx gateway with FXO ports. The device must be running on the correct firmware version. To determine this, click the **Home** button and make sure the gateway has version 6.60A.265.010.
- MaxCS must already be configured behind NAT, including the Port Forwarding and MaxCS Enterprise Manager settings. AltiGen IP phones behind NAT must already be properly working.
- The EdgeMarc SBC's WAN address, 10.40.0.95, must be listed in MaxCS under "Trusted SIP Device List." If it is not listed there, it will be treated as a malicious SIP device due to excessive SIP messages from the address.
- All AltiGen IP phones must be on firmware 2xA2 or later.
- If you are using Polycom IP phones, then they must be configured correctly. For instructions, refer to AltiGen's *Polycom IP Phone Configuration Guide*.
- The EdgeMarc SBC must already be running, with a public IP address. Computers behind the EdgeMarc SBC must already be able to browse the Internet through the EdgeMarc SBC.
- Your AltiGen IP phones and any Polycom phones must already be able to work with MaxCS behind the EdgeMarc SBC.
- You must have all necessary valid EdgeMarc SBC licenses, such as Licensed Calls, SIP Support, and SIP Survivability.

Voice Quality between IP Phones behind NAT

When two IP phones connect behind the same NAT, if the voice codec is the same and if there are no recording or silent monitoring features enabled on these two IP phones, the voice stream will go peer-to-peer. When voice streams go peer-to-peer, voice latency is reduced; this improves the voice quality.

If voice streams cannot go peer-to-peer due to the conditions mentioned above, voice streams will go back to the MaxCS server.

To verify that such calls transmit voice streams peer-to-peer, check the following:

1. Make sure that there are no active calls in the MaxCS system.
2. Place a call between two IP phones behind the same NAT and make sure that the voice stream is connected both directions. **Do not hang up the call**.
3. In MaxAdministrator, select **View** > **Current Resource Statistics**.

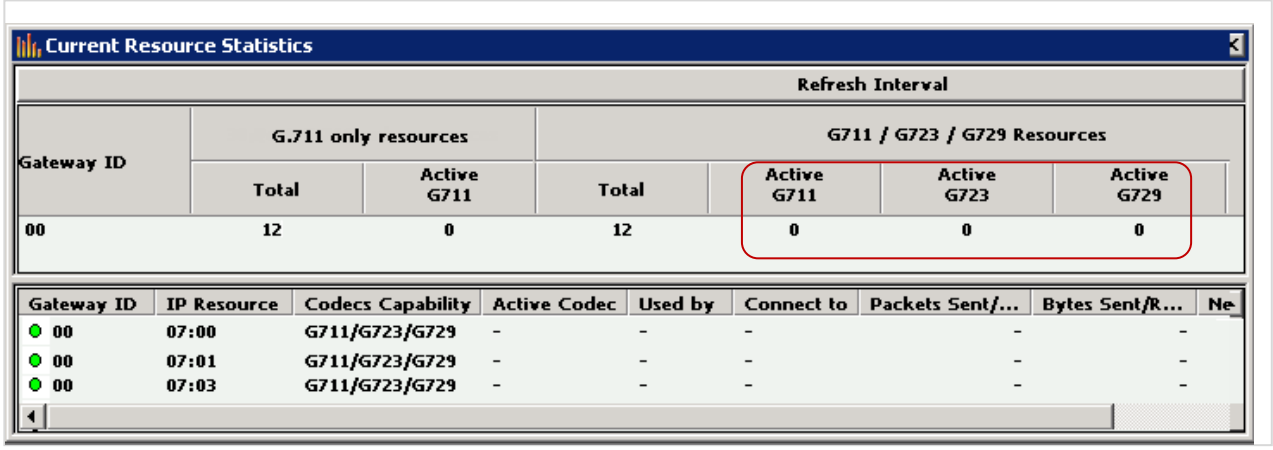


Figure 2: The Current Resource Statistics panel

If your call is not a peer-to-peer connection, one or more of "Active G711, G723, or G729" will be a non-zero value. If your call is a peer-to-peer, all of these values will be 0, as shown in Figure 2.

EdgeMarc Survivability Configuration

This section shows how to configure your EdgeMarc SBC for PSTN survivability.

Note: If you do not follow these configuration steps in EdgeMarc, then your phone may still work. However, you will not be able to use some features provided by the EdgeMarc device, including the traffic shaping, monitoring, and survivability features.

1. Log into your EdgeMarc SBC configuration page.
2. From the menu on the left, select **VoIP**. Clear the checkbox *Allow Shared Usernames*.

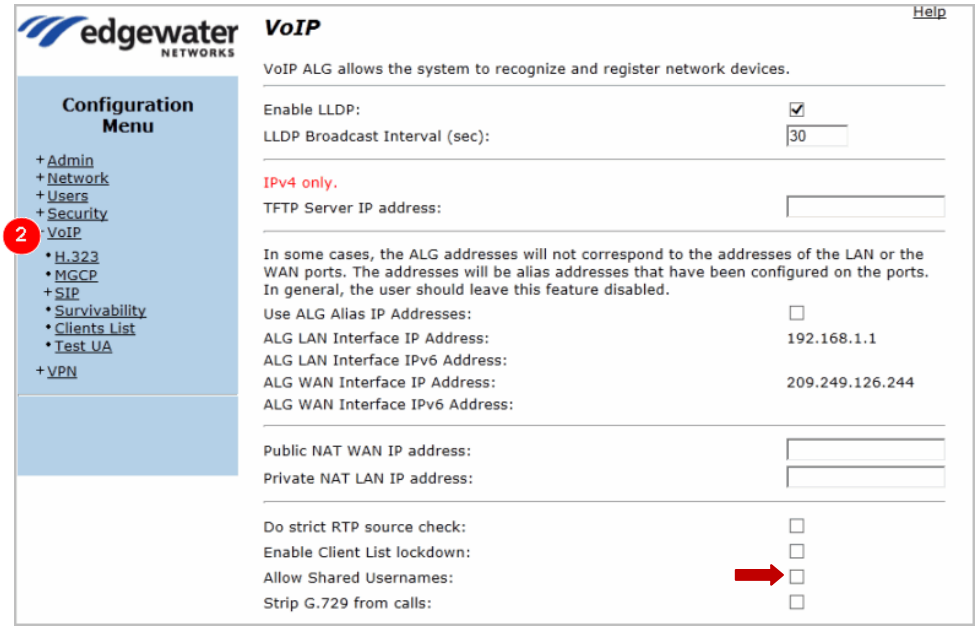


Figure 3: Clearing the "Allow Shared Usernames" checkbox

3. Select **VoIP** > **SIP**. On the SIP Settings page, click **Create**.

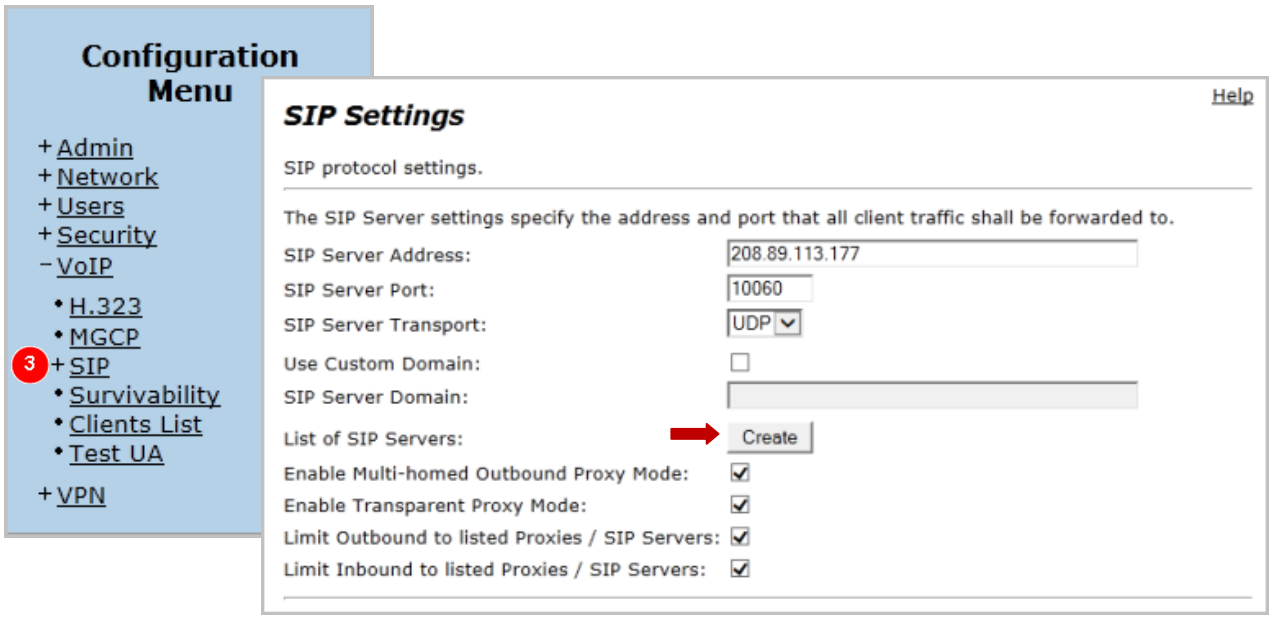


Figure 4: Click "Create" to add a new proxy

4. In **Add a new proxy**, add the MaxCS IP address and the port (in our example, this is 10060). This is usually a public IP address. In this example, we will use the IP address 10.40.1.43.

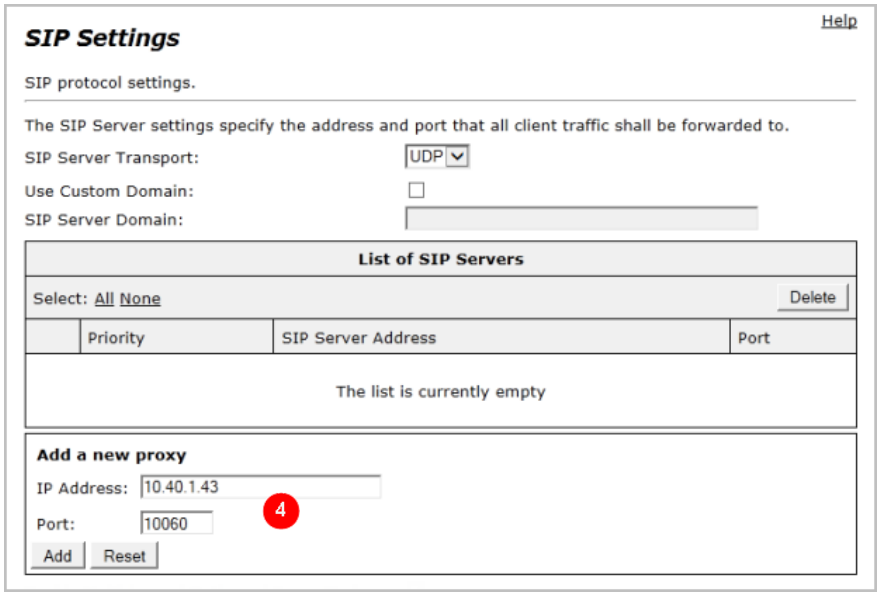


Figure 5: Add a new proxy with the MaxCS IP address and port

5. On the same page, update the UDP and TCP settings:

- Set *Client Listening Port(s)* to **5060, 10060**
- Set *Server Facing Port* to **5060**
- Set the TCP *Port* to **5060**

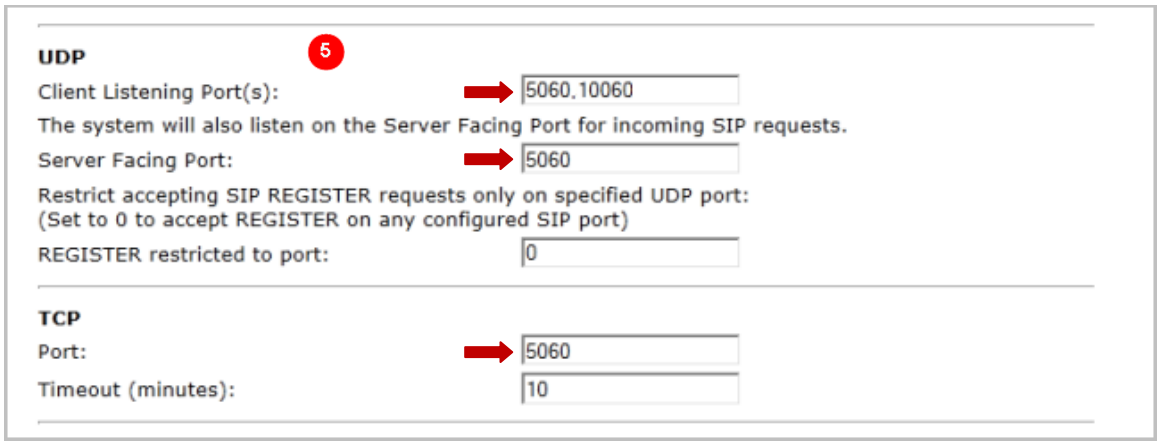


Figure 6: Configure UDP and TCP settings

6. Select **VoIP** > **Survivability**. Under the *Common Settings* section, make sure that *Survivability* is set to **Enabled (auto)**. (Your EdgeMarc device must have a SIP Survivability license, as mentioned in the *Prerequisites* section on page 5.)

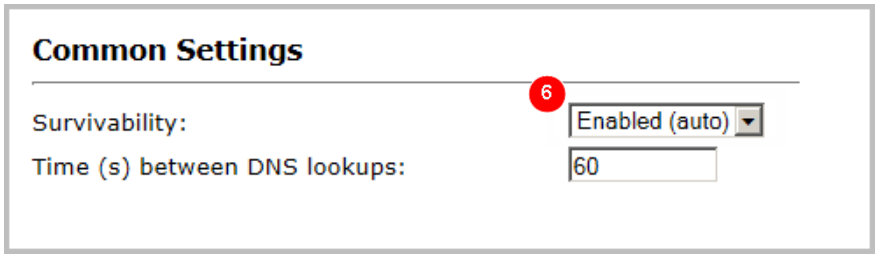


Figure 7: Check that Survivability is enabled

7. Select **Admin** > **Registration Status**, click the License key to view license details. Confirm that *SIP Survivability* shows as **on**.

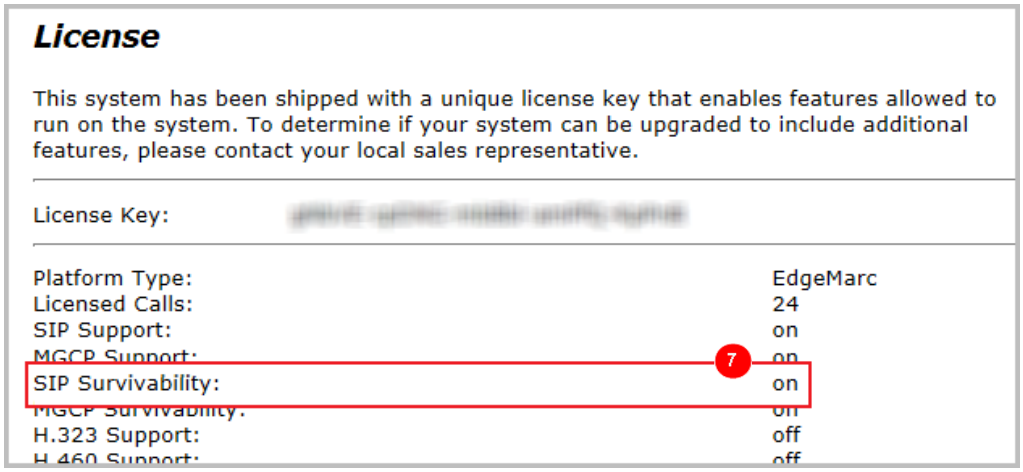


Figure 8: Check license details

8. Select **VoIP** > **Survivability**. In the *SIP Registration Control* section, update the following settings:

- Clear the *Enable Phone Expires Override* checkbox
- Clear the *Enable Soft-Switch Expires Override* checkbox
- Set *Rate-Pacing behavior* to **Non (Rate-pacing is disabled)**

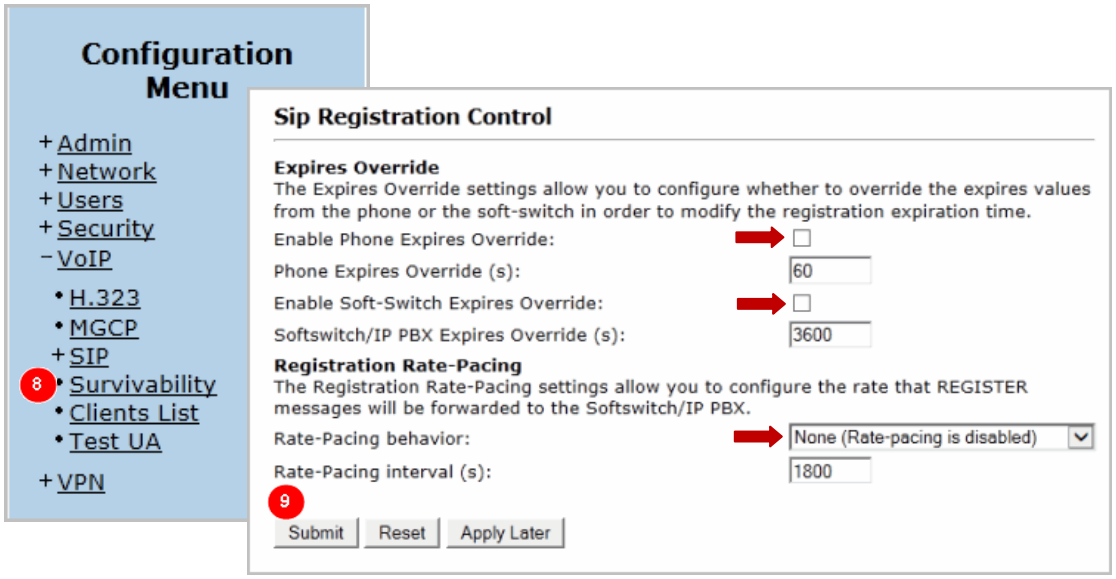


Figure 9: Configure the SIP Registration settings

9. Click **Submit**. Wait a few minutes while the EdgeMarc SBC restarts.

Verification

After the EdgeMarc SBC restarts, verify the settings.

1. Log in and select **VoIP** > **Survivability**. Wait a few moments; the small dot on the left should change to green. (Refresh your browser as needed.) If the dot stays red after several minutes, the configuration is not correct – go through the steps in the previous section to check each setting.

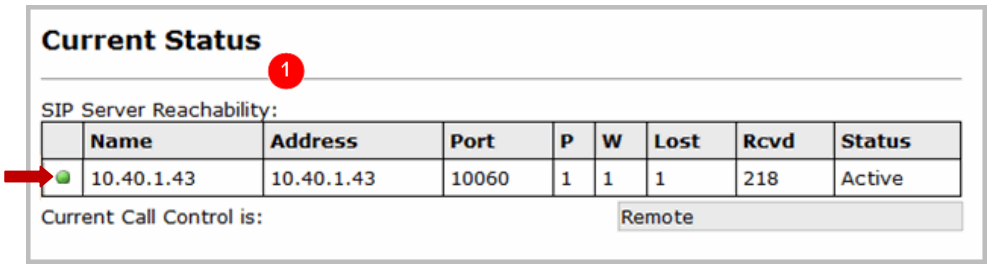


Figure 10: Check that the status changes to Active

2. Select **Admin** > **System Information**. The *Number of Active Streams* value should be 0.

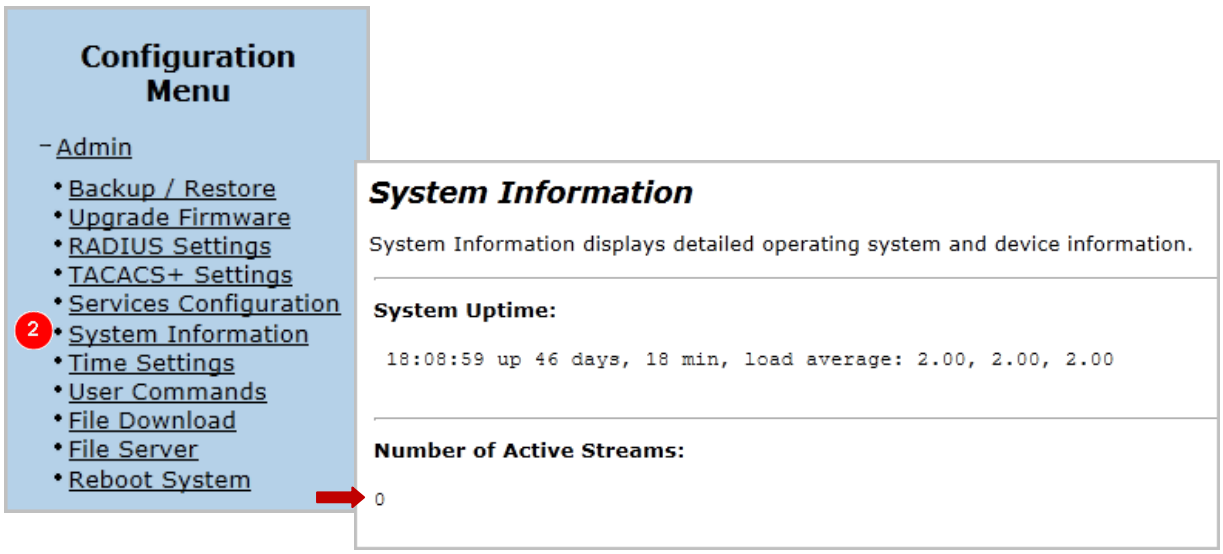


Figure 11: Check that the number of active streams is zero

3. Pick up an AltiGen IP phone that is behind the EdgeMarc SBC. Dial **# #** to enter voicemail. While the phone is playing back voicemail, refresh the browser. The *Number of Active Streams* value should now be **1**.

EdgeMarc Survivability Configuration via AudioCodes FXO Ports

This section describes how to implement survivability by using an EdgeMarc SBC with an AudioCodes gateway that has FXO ports.

EdgeMarc Configuration

1. Log into the EdgeMarc configuration page and select **VoIP** > **SIP** > **ALG**. In the *SIP Trunking Devices* section, add a device with the following configuration:

- Set *Action* to **Add new trunking device**
- For *Name* enter **audiocodes gw**
- For *Address* enter **192.168.1.20**
- For *Port* enter **10060**

2. Click **Commit**.

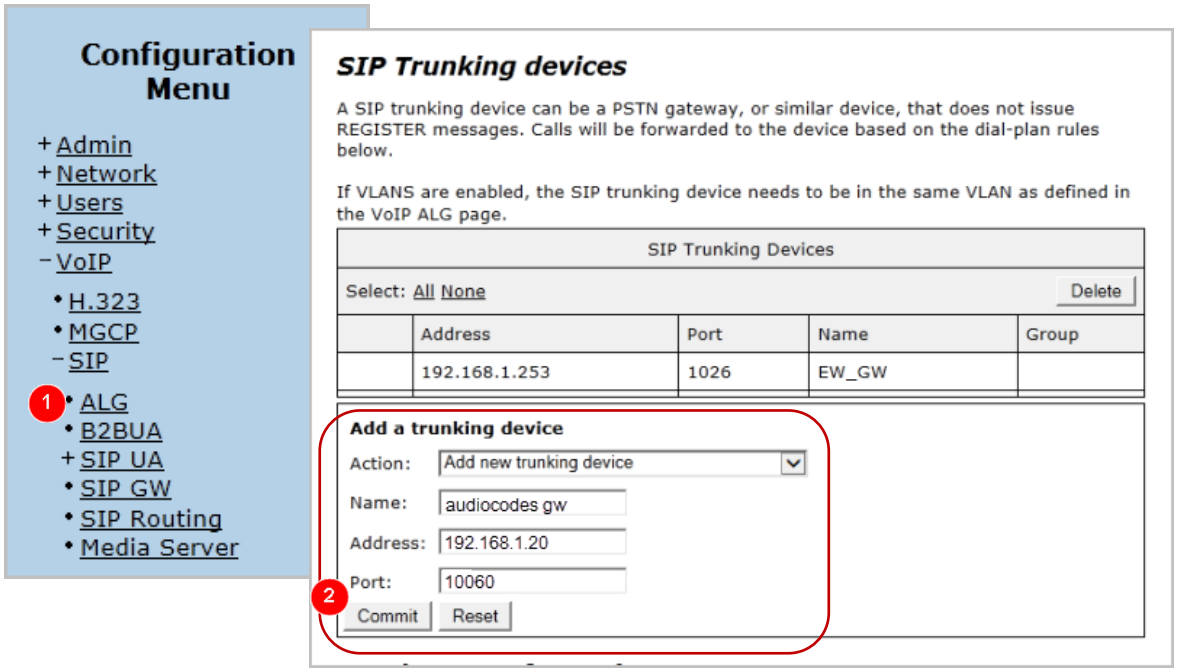


Figure 12: Add a new trunking device

3. On the same page, under the *Dialing Rules* section, add a new rule with the following settings:

- Set *Action* to **Add new rule**
- Set *Type* as **Inbound**
- Check the *Default Rule* checkbox
- Set *Strip digits* to **1** (In survival mode, a user still dials trunk access code '9' when making a PSTN call. By setting this to 1, the '9' will be stripped before sending it to the AudioCodes gateway.)
- Set *Trunking device* to **audiocodes gw (192.168.1.20:10060)**

4. Click **Commit**.

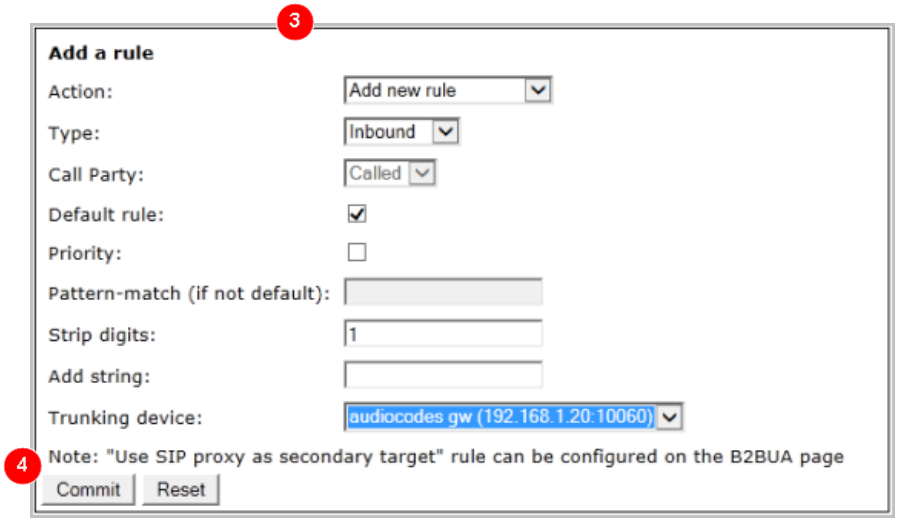


Figure 13: Add a new rule

5. At this point, while in survival mode a user would have to dial 9-911 to make an emergency call. To enable 911 dialing without dialing an additional digit 9, add another rule:

- Set *Action* to **Add new rule**
- Set *Type* as **Inbound**
- For *Pattern Match (if not default)* enter **911**
- Set *Strip digits* to **0**
- Set *Trunking device* to **audiocodes gw (192.168.1.20:10060)**

6. Click **Commit**.

AudioCodes MP-118 Configuration

This section describes how to implement survivability by using an EdgeMarc SBC with an AudioCodes MP118 gateway.

If you have an MP118 gateway that you've already configured for FXS analog support, you can use that same gateway to implement FXO survivability. If this is the case, you may have already performed some of the steps in this section.

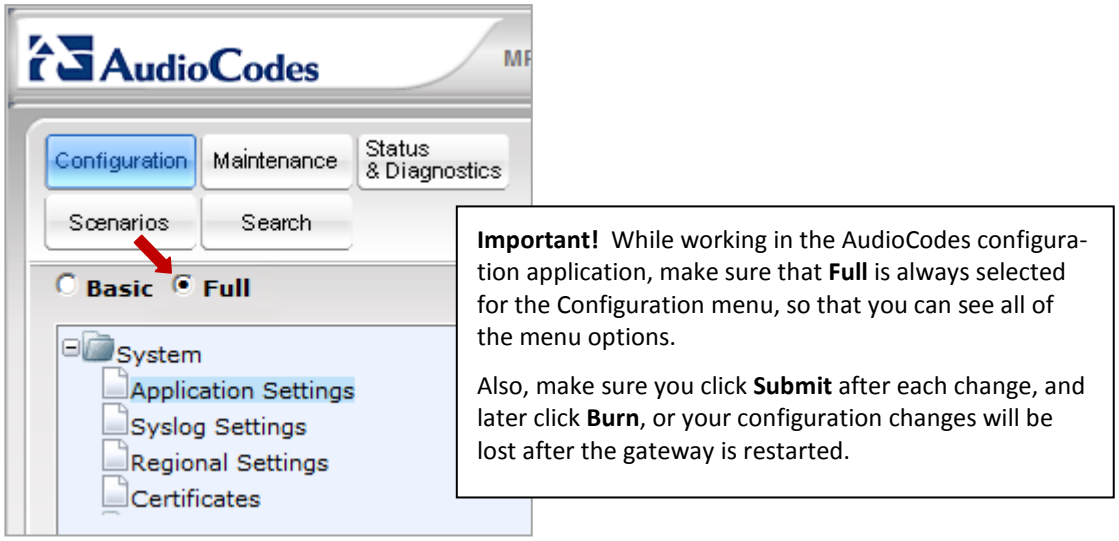


Figure 14: Choose Full for the AudioCodes Configuration menu

1. Log into the AudioCodes configuration tool. Select **VoIP** > **Network** > **IP Interfaces Table** and confirm that the IP address is 192.168.1.20 (the AudioCodes IP address from Figure 1) and that the prefix length and the network's gateway IP addresses are set up properly. (This is a read-only page; the GUI may look slightly different from the figure.)

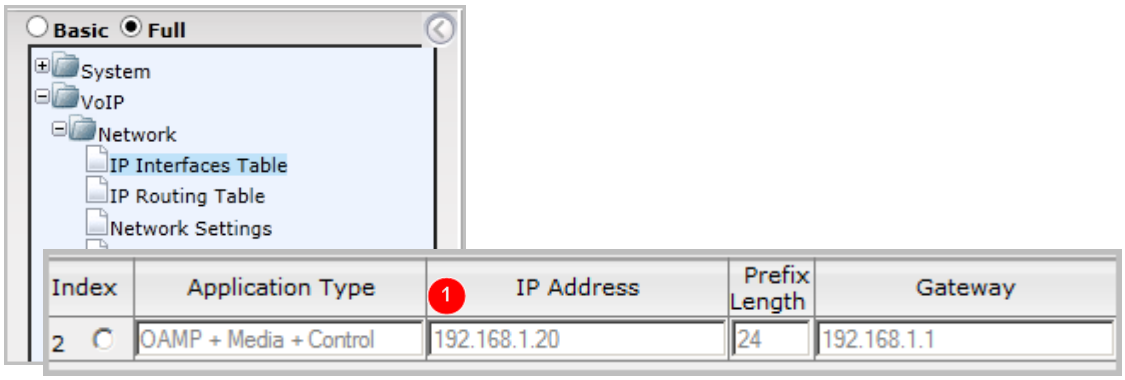
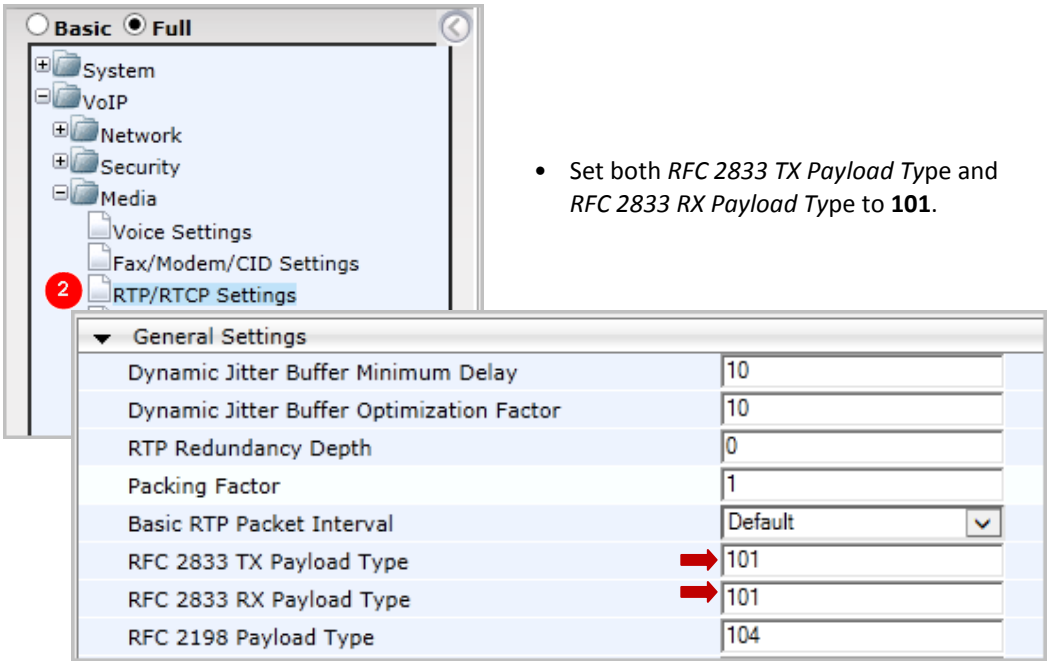


Figure 15: Check the network IP settings

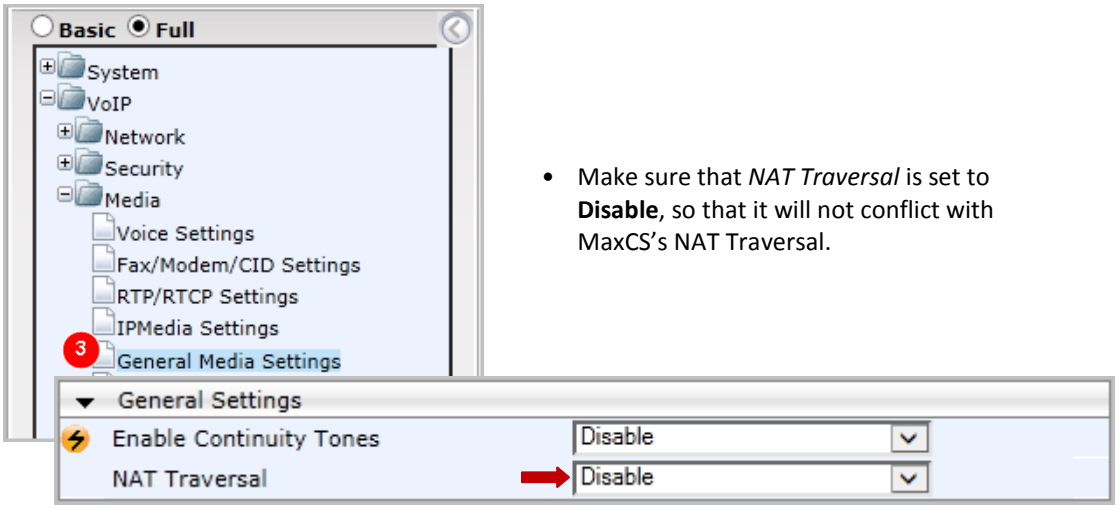
2. Select **VoIP** > **Media** > **RTP/RTCP Settings**.



- Set both *RFC 2833 TX Payload Type* and *RFC 2833 RX Payload Type* to **101**.

Figure 16: Configure the Media RFC 2833 Payload settings

3. Select **VoIP** > **Media** > **General Media Settings**.



- Make sure that *NAT Traversal* is set to **Disable**, so that it will not conflict with MaxCS's NAT Traversal.

Figure 17: Set NAT Traversal to Disable

4. Select **Control Network** > **Proxy Sets Table**. Configure *Proxy Set ID 0* as follows:

- For *Proxy Address 1* enter **10.40.1.43** (the MaxCS public IP address). Set its *Transport Type* to **UDP**.
- Set *Enable Proxy Keep Alive* to **Disable**.

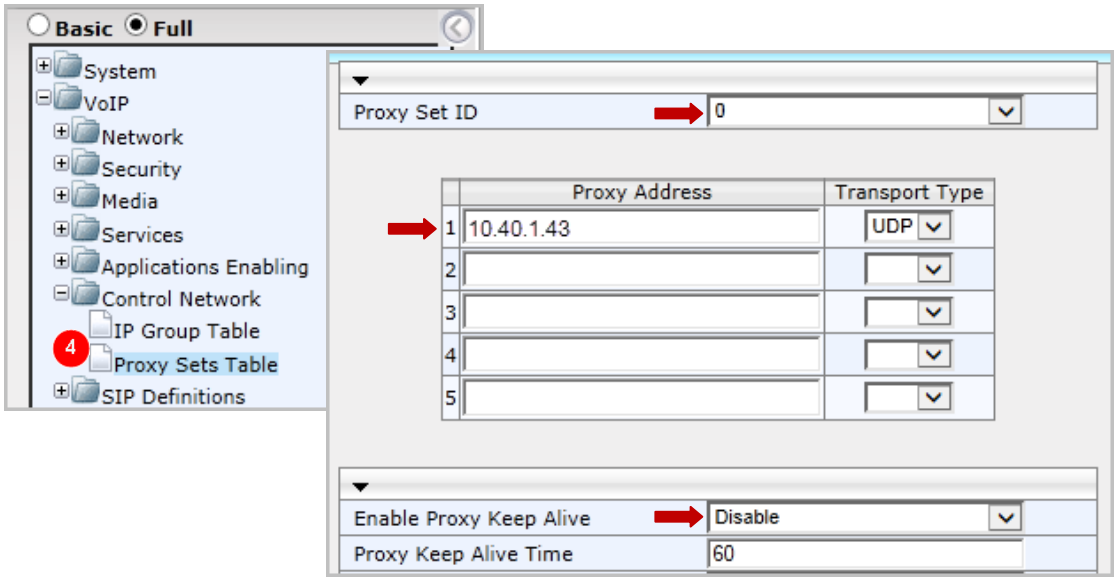
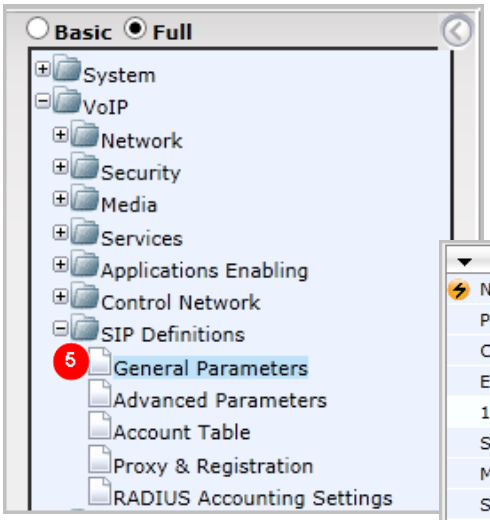


Figure 18: Set Proxy Set 0 with the MaxCS public IP address

5. Select **VoIP** > **SIP Definitions** > **General Parameters**.



- Make sure that *NAT IP Address* is **0.0.0.0**.
- Set *Enable Early Media* to **Disable**.
- Set *SIP Transport Type* to **UDP**.
- Set *SIP UDP Local Port* to **10060**.
- Set *SIP Destination Port* to **10060**.

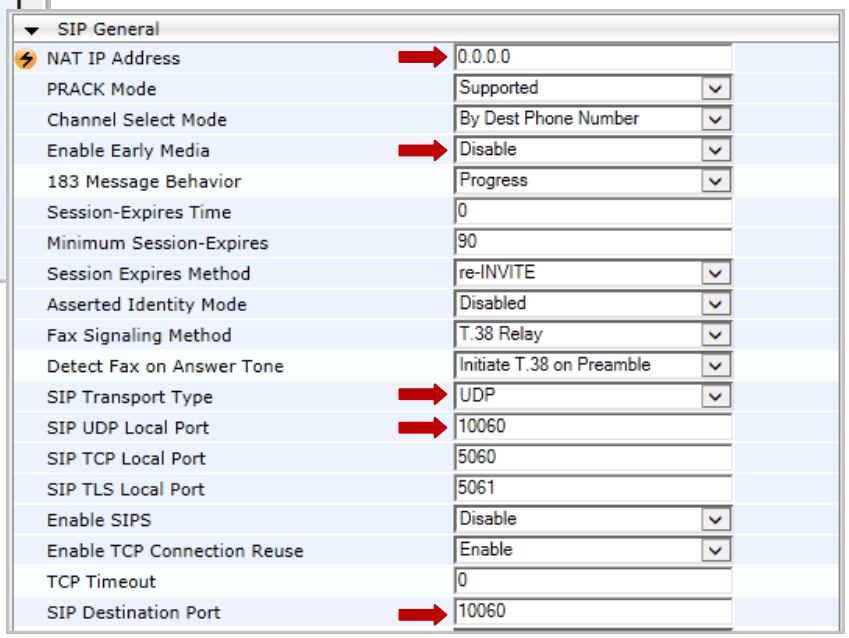
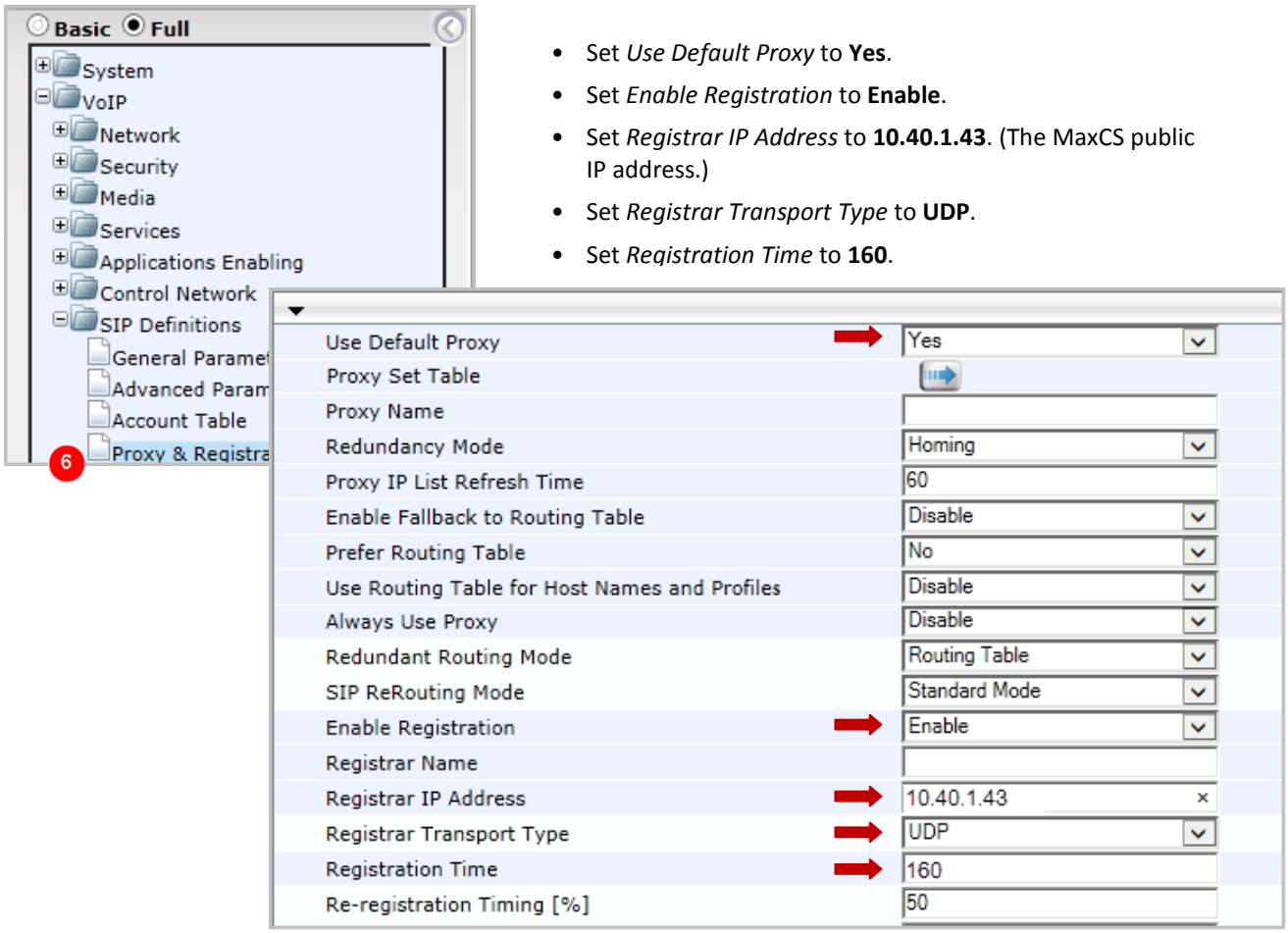


Figure 19: Set SIP general parameters

6. Select **VoIP** > **SIP Definitions** > **Proxy & Registration**.



- Set *Use Default Proxy* to **Yes**.
- Set *Enable Registration* to **Enable**.
- Set *Registrar IP Address* to **10.40.1.43**. (The MaxCS public IP address.)
- Set *Registrar Transport Type* to **UDP**.
- Set *Registration Time* to **160**.

Figure 20: Configure proxy and registration settings

7. Select **VoIP** > **Coders and profiles** > **Coders**. Make sure that both G.711U-law and G.729 are in the list.

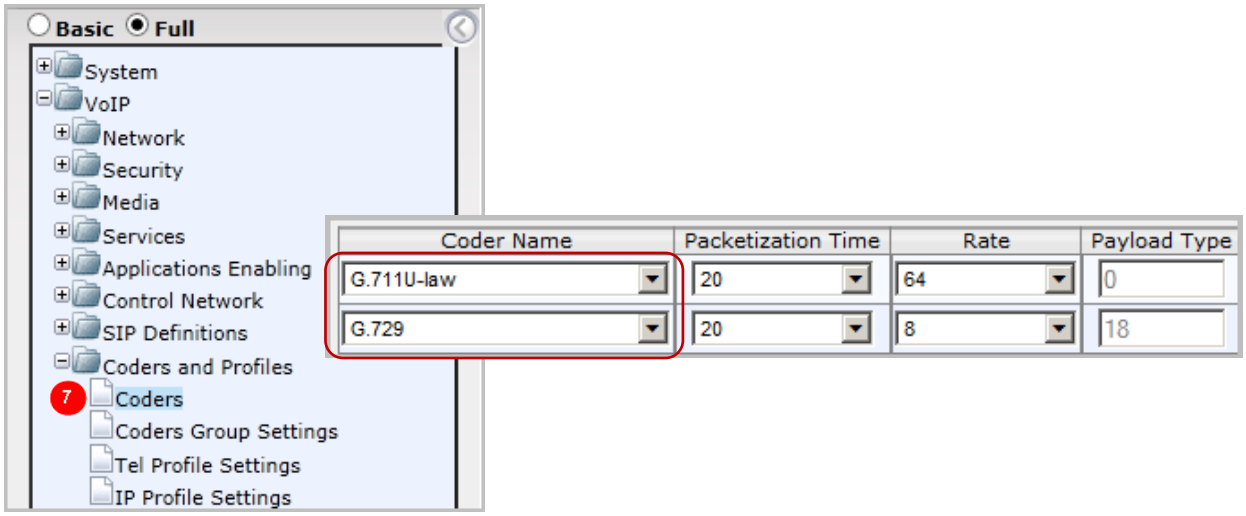
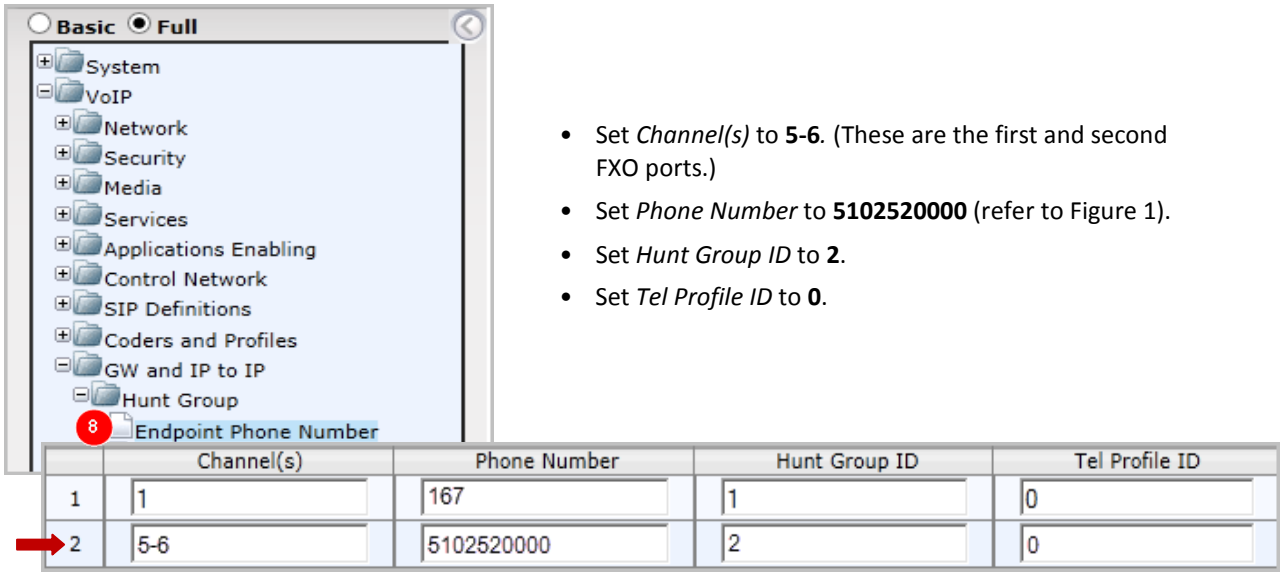


Figure 21: Confirm that G.711U-law and G.729 are listed in the Coders and Profiles page

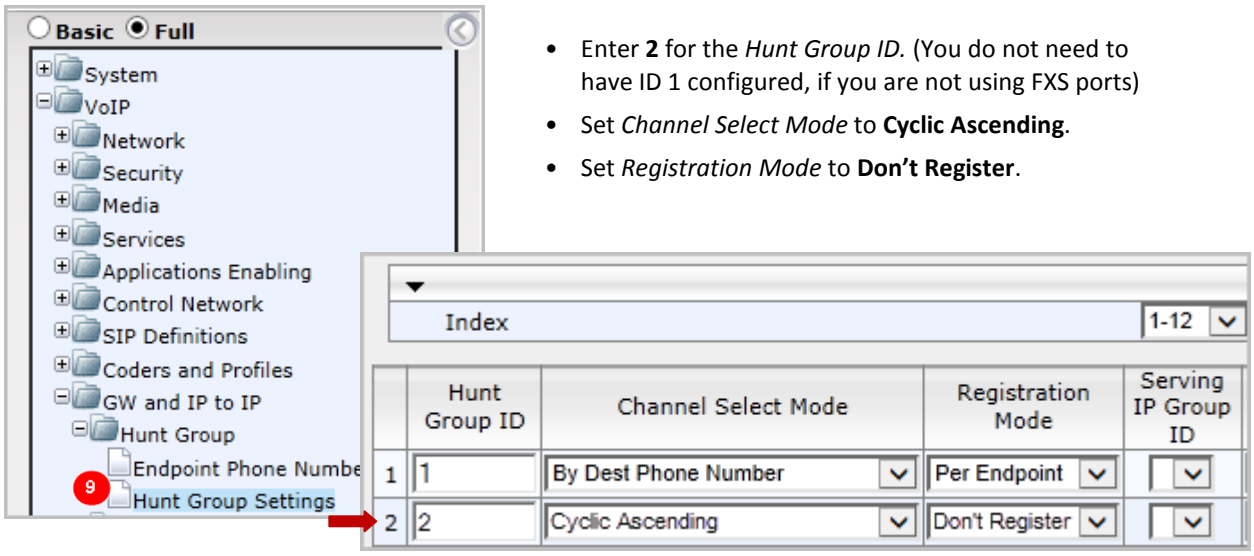
8. Select **VoIP** > **GW and IP to IP** > **Hunt Group** > **Endpoint Phone Number**. Select any available entry and configure it as follows (the example shown uses Entry 2):



- Set *Channel(s)* to **5-6**. (These are the first and second FXO ports.)
- Set *Phone Number* to **5102520000** (refer to Figure 1).
- Set *Hunt Group ID* to **2**.
- Set *Tel Profile ID* to **0**.

Figure 22: Configure the hunt group endpoint phone number

9. Select **VoIP** > **GW and IP to IP** > **Hunt Group** > **Hunt Group Settings**. Select any available entry and configure it as follows (this example uses Entry 2):



- Enter **2** for the *Hunt Group ID.* (You do not need to have ID 1 configured, if you are not using FXS ports)
- Set *Channel Select Mode* to **Cyclic Ascending**.
- Set *Registration Mode* to **Don't Register**.

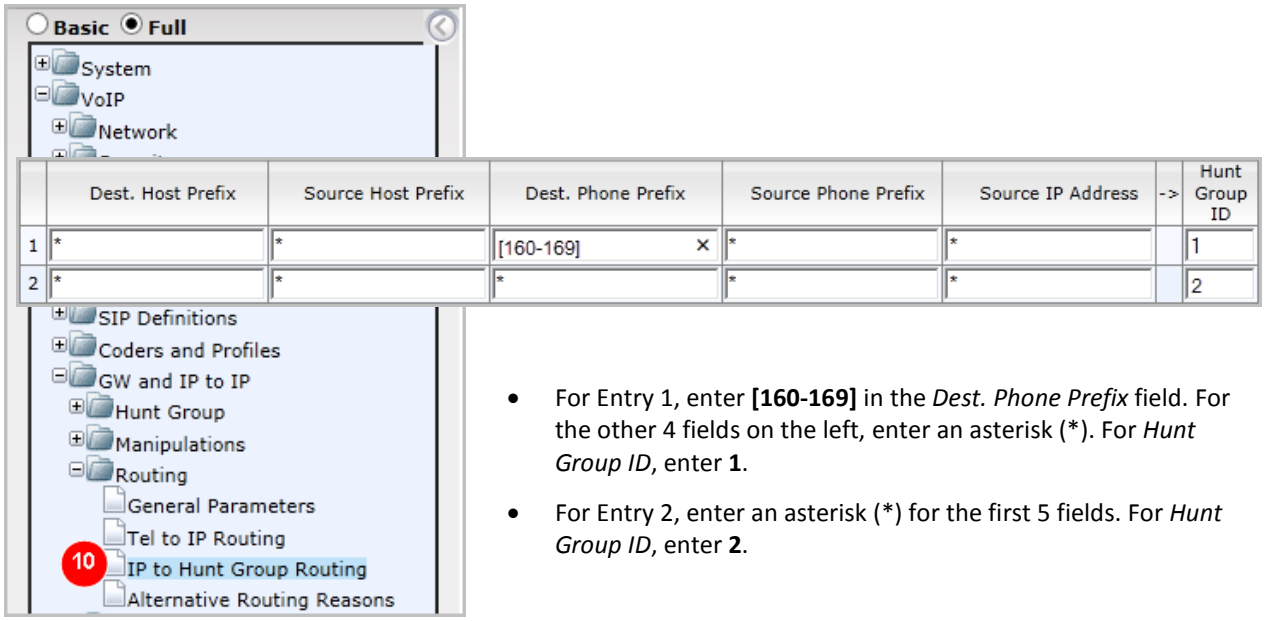
Figure 23: Configure hunt group settings

Note: Entry 1 in this figure is a configured FXS port not covered in this guide.

10. Select **VoIP** > **GW and IP to IP** > **Routing** > **IP to Hunt Group Routing**.

If there are no analog extensions behind this MP118 gateway, move the second row's content to the first row (all row 1 fields should contain an asterisk), make sure that the *Hunt Group ID* for row 1 is set to **2**, and skip ahead to the next step (step 11 on page 19).

If there are analog extensions behind this gateway, you must make changes. In our example, with extensions 160 - 169 behind the gateway FXS ports, the following settings are required:



- For Entry 1, enter **[160-169]** in the *Dest. Phone Prefix* field. For the other 4 fields on the left, enter an asterisk (*). For *Hunt Group ID*, enter **1**.
- For Entry 2, enter an asterisk (*) for the first 5 fields. For *Hunt Group ID*, enter **2**.

Figure 24: Configure IP to hunt group routing rules

11. Select **VoIP** > **GW and IP to IP** > **DTMF and Supplementary** > **DTMF & Dialing**.

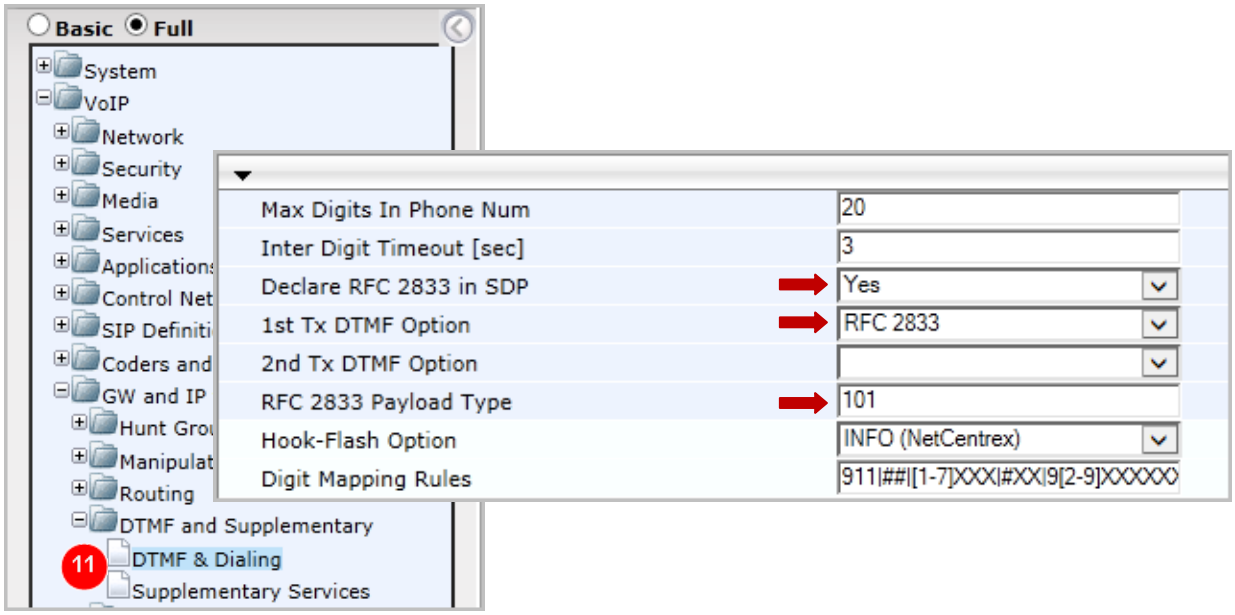


Figure 25: Configure dialing settings

12. Select **VoIP** > **GW and IP to IP** > **DTMF and Supplementary** > **Supplementary Services**.

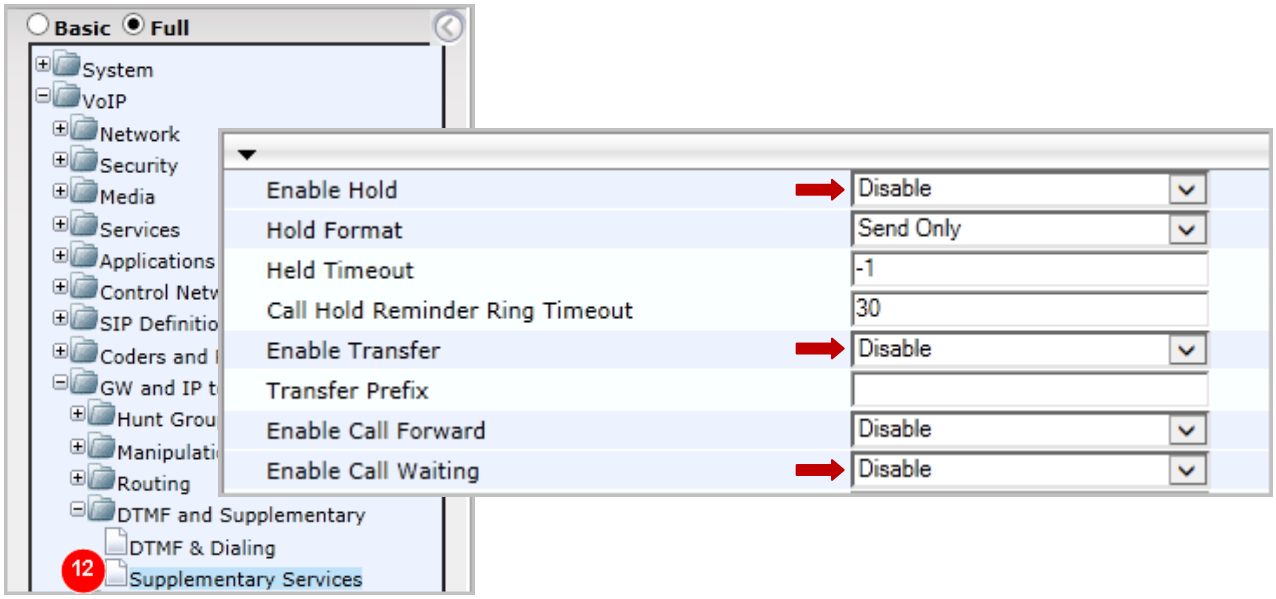
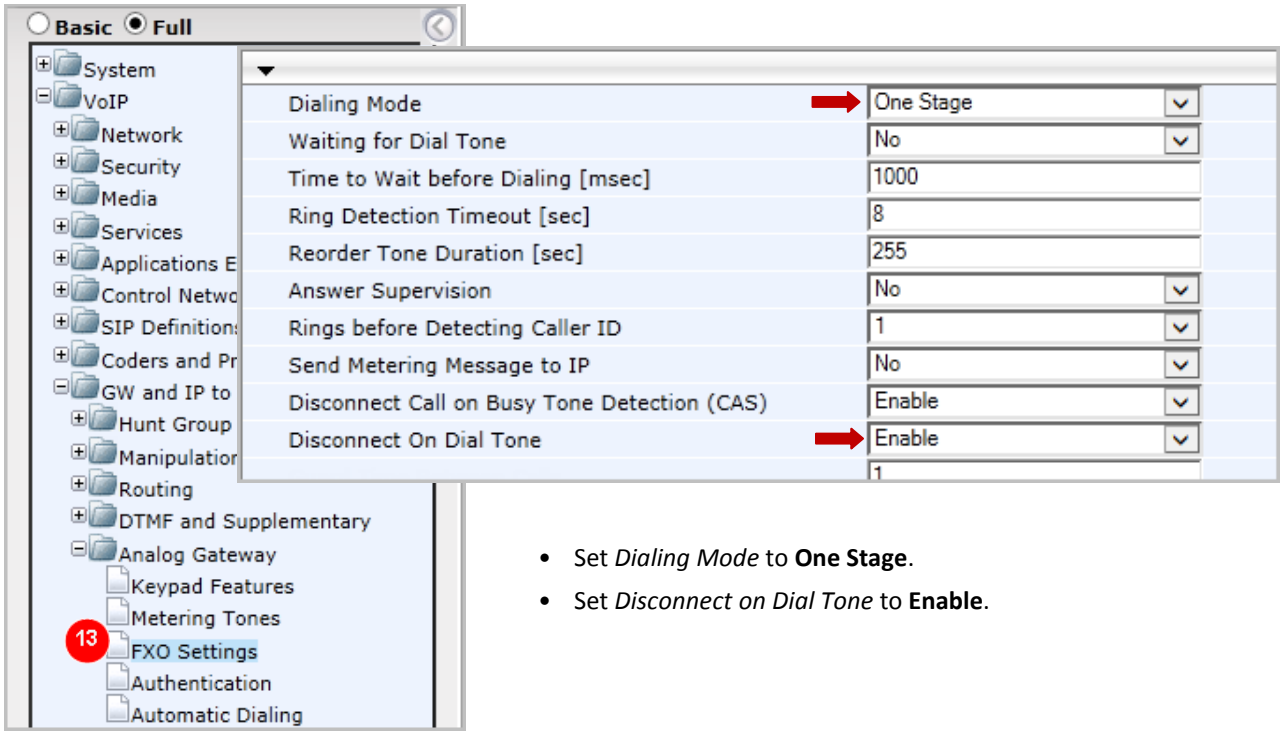


Figure 26: Disable the Hold, Transfer, and Call Waiting features

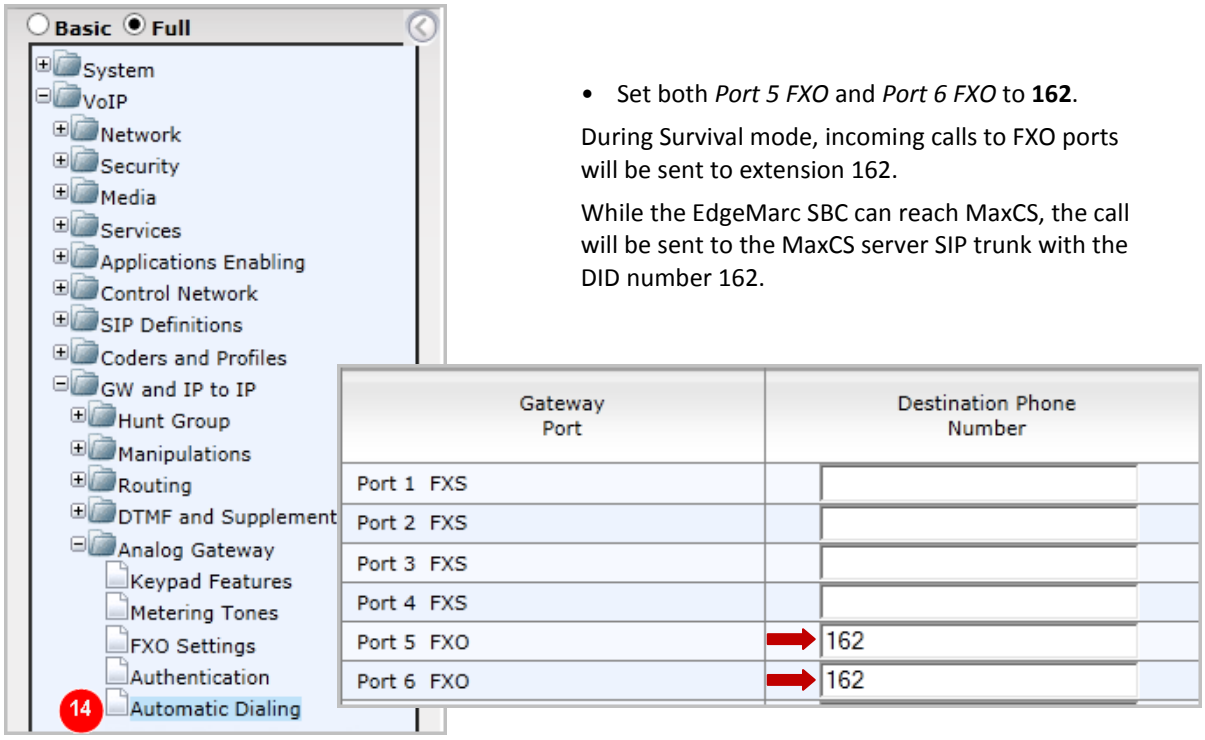
13. Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **FXO Settings**.



- Set *Dialing Mode* to **One Stage**.
- Set *Disconnect on Dial Tone* to **Enable**.

Figure 27: Configure FXO settings

14. Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **Automatic Dialing**.



- Set both *Port 5 FXO* and *Port 6 FXO* to **162**.

During Survival mode, incoming calls to FXO ports will be sent to extension 162.

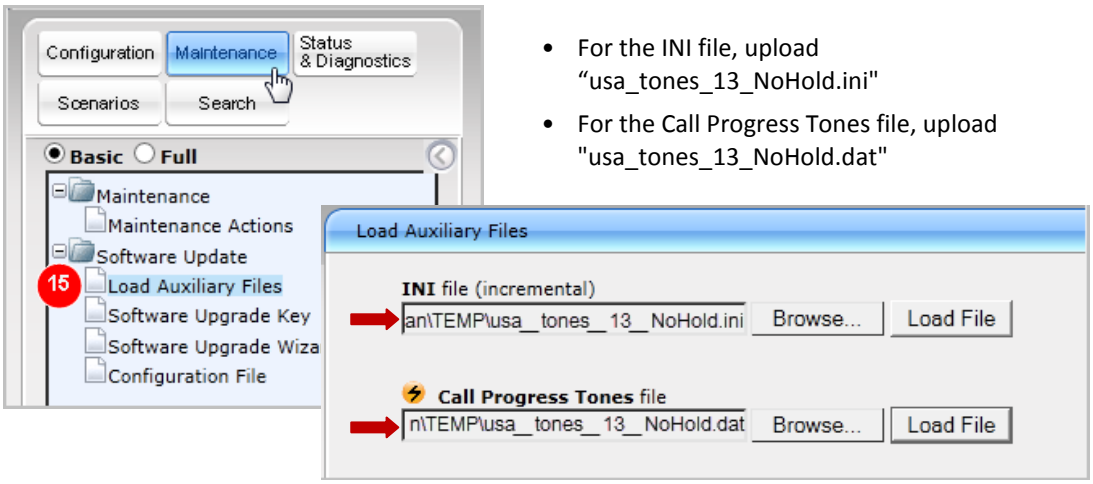
While the EdgeMarc SBC can reach MaxCS, the call will be sent to the MaxCS server SIP trunk with the DID number 162.

Figure 28: Route incoming calls to FXO ports to extension 162

15. At this point, you will upload the two files that came with this configuration guide, as mentioned on page 3.
    - usa_tones_13_NoHold.ini
    - usa_tones_13_NoHold.dat

    If you do not load these files, callers may hear an on-hold beep tone while calls are being connected or disconnected.

    Click the **Maintenance** button above the menu. Then select **Software Update** > **Load Auxiliary Files**.



- For the INI file, upload "usa_tones_13_NoHold.ini"
- For the Call Progress Tones file, upload "usa_tones_13_NoHold.dat"

Figure 29: Upload the .INI and .DAT files

Note: If you already loaded these files while configuring analog extensions on this MP1xx gateway, you do not need to reload these files again.

16. Submit this last change. On the toolbar, click **Burn**. Restart the AudioCodes gateway. Continue to the next section to adjust settings in your AltiGen IP phones.

Figure 30: Click the Burn button to save this configuration

AltiGen IP Phone Configuration

If you are using 3rd party IP phones on FXS ports behind the AudioCodes gateway, those phones need no additional changes for survivability.

As mentioned earlier, AltiGen IP phones should already be working properly behind the firewall/NAT. Therefore, extension 161(in Figure 1) should work correctly. The AltiGen IP phones will need a few configuration changes to implement survivability.

1. On the AltiGen IP phone, press the **Menu** button.

2. In the phone's menus, navigate to **System** > **Emergency Gateway**. Enter the MaxCS IP address (in our example, this IP address is 10.40.1.43).
3. In the phone's menus, navigate to **System** > **Enable SIP Reg**. Set the value to **Yes**.

Tip: For AltiGen IP phones, if the phone has no voice path in survival mode, look in the phone's menus for the setting **Network** > **Enable NAT**. If your phone has that setting, change the value to **No**.

Verification

Now that you have configured the phones and the gateway, verify that the phones work correctly.

1. Log into EdgeMarc SBC configuration page and select **VoIP** > **Survivability**. Wait a few moments; the small dot on the left should change to green. (Refresh your browser as needed.)

   If the dot stays red after several minutes, it means that your configuration is not correct – go through the steps in the previous sections and check each setting.

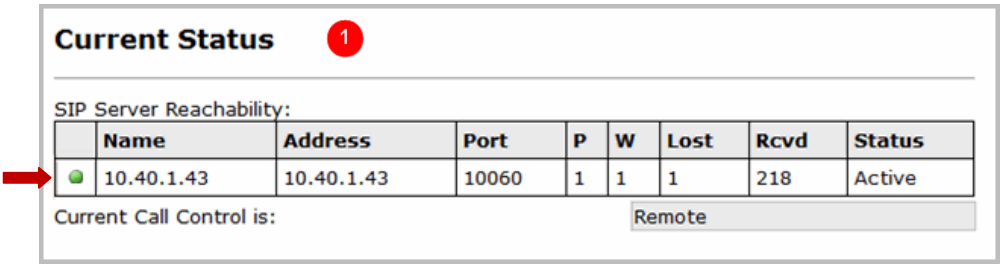


Figure 31: Check that the status changes to Active

2. To emulate a loss of network connection to the MaxCS server, open MaxCS MaxAdministrator *Boards* view. Double click the **SIPSP** entry, select **Board Configuration**, and click **Advanced Configuration**.

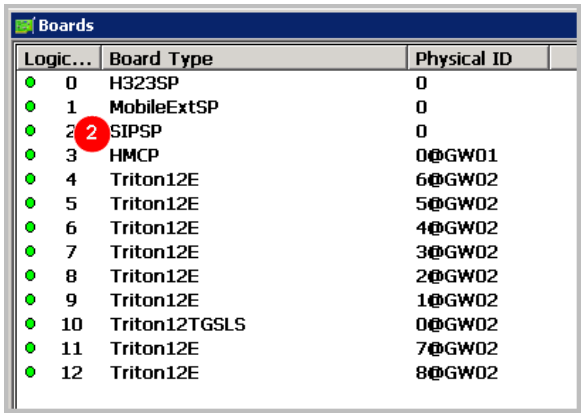
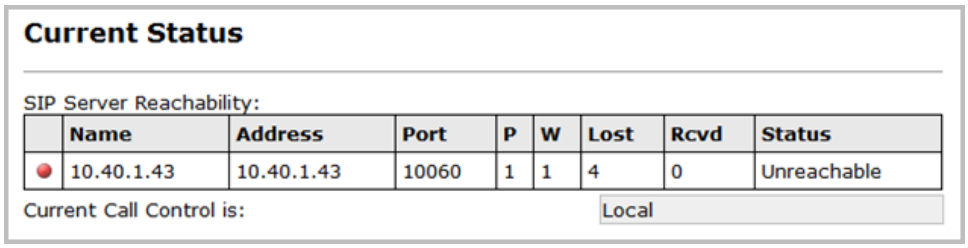


Figure 32: In Boards view, open the Advanced Configuration panel for SIPSP

3. Add the EdgeMarc SBC's WAN address (in our example, this is 10.40.0.95) to the Malicious SIP Device List. You do this so that MaxCS will not respond to any SIP messages from that IP address.



Current Status

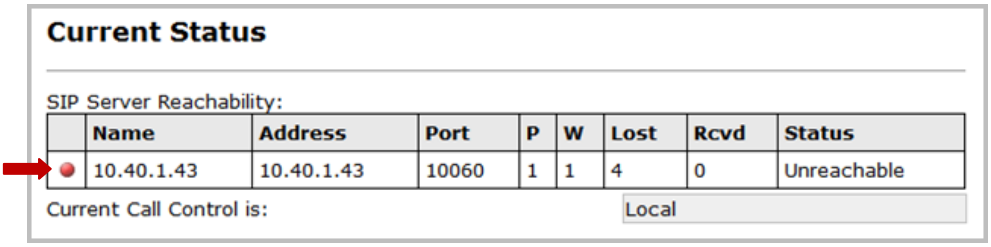
SIP Server Reachability:

	Name	Address	Port	P	W	Lost	Rcvd	Status
●	10.40.1.43	10.40.1.43	10060	1	1	4	0	Unreachable

Current Call Control is: Local

Figure 33: Add the EdgeMarc's WAN address to the Malicious SIP Device list

4. Return to the EdgeMarc SBC *Current Status* page and refresh the page. You should now see a red dot instead of a green one, and the status should say *Unreachable.*



Current Status

SIP Server Reachability:

	Name	Address	Port	P	W	Lost	Rcvd	Status
●	10.40.1.43	10.40.1.43	10060	1	1	4	0	Unreachable

Current Call Control is: Local

Figure 34: The status in the EdgeMarc page should change to "Unreachable"

5. For an AltiGen IP phone, such as extension161 in our example, make a change that will simulate a loss of connection to MaxCS: On the IP phone, press **#27**, enter **161** as the extension number, and enter an incorrect password.

   A minute after you do this, the phone will register to the EdgeMarc SBC and will show *LOCAL* on the phone's display.

6. In the AudioCodes configuration tool, set *Automatic Dialing* to extension 161 instead of 162: Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **Automatic Dialing**. (Refer to Figure 28.)

   From your mobile phone, make an inbound call to the FXO port (5102520000). The call should be sent to the AltiGen IP phone extension 161. From your mobile phone, make an inbound call to the FXO port (5102520000). The call should be sent to the AltiGen IP phone extension 161.

7. Use an AltiGen IP phone to make an outbound PSTN call starting with trunk access code (for example, 915102520004. The call should be sent through the FXO ports on the AudioCodes gateway. Trunk access code '9' will be removed before the number sent to the FXO ports.

   Pick up the call and verify that you have a two-way connection.

8. Remove 10.40.0.95 from the Malicious SIP Gateway List (see Figure 33). Add 10.40.0.95 back to the Trusted SIP Gateway List (In Boards view, double click the **SIPSP** entry, select **Board Configuration**, and click **Advanced Configuration**). . Use #27 to log into AltiGen IP extension 161 with the correct password.

   Wait a few minutes; the phone system should be recovered and the EdgeMarc *Current Status* should once again show a green dot.

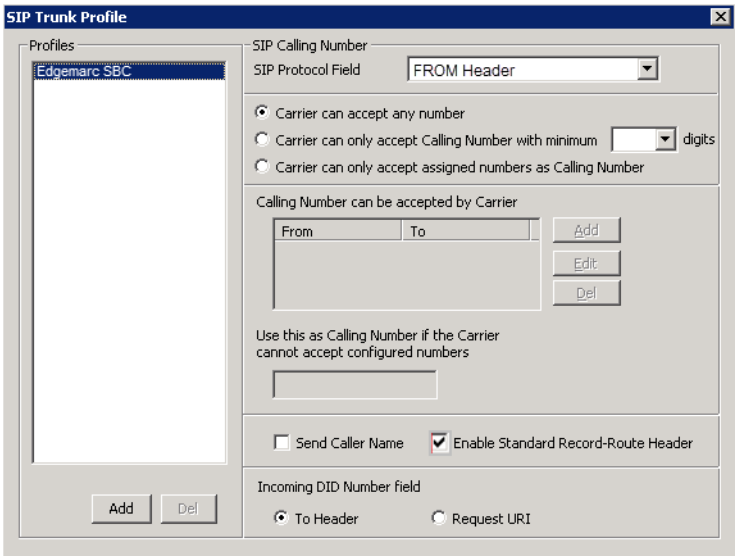
MaxCS Configuration

Under normal conditions, when someone makes a call to 5102520000 through PSTN, the AudioCodes gateway's FXO port receives the call and sends it to MaxCS (10.40.1.43) as a SIP tie-trunk call with a called number162 (as shown in Figure 1).

In this case, the Polycom IP phone 162 will ring. When the call is answered, however, the caller ID on the Polycom IP phone shows the EdgeMarc SBC's IP address.

It is better to route incoming calls from the AudioCodes FXO ports to a workgroup or some other destination instead of routing them to extension 162. To achieve this, you can set up SIP trunks instead of SIP tie-trunk to handle calls.

1. Log into MaxCS MaxAdministrator and open Trunk view.
2. Double-click SIP trunk **03:0100**, click **Trunk Properties**. In the *SIP Trunk Configuration* panel, click **SIP Trunk Profile**.
3. Click **Add**. Add a new profile with the following settings:

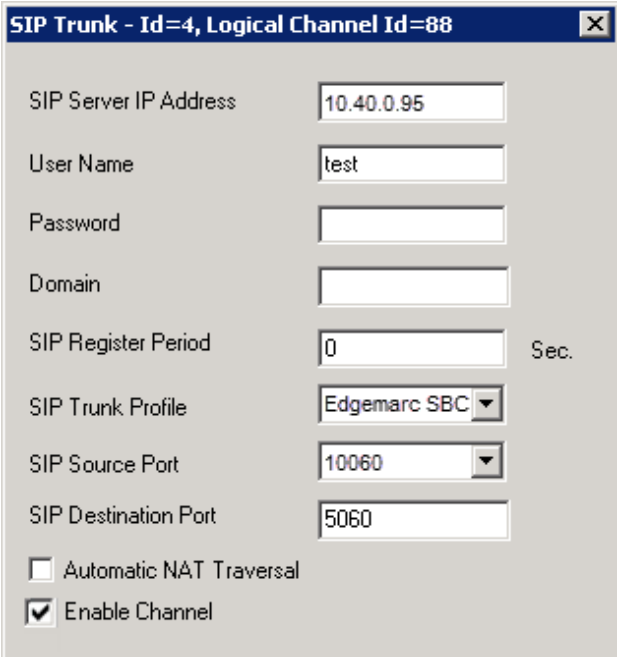


- Enter the name **EdgeMarc SBC**
- Set *SIP Protocol Field* to **FROM Header**
- Check the *Enable Standard Record Route Header* box

Figure 35: Add a new SIP Trunk profile

4. Return to Trunk view. Double-click SIP trunk **03:0100** and click **Trunk Properties**.

5. Edit channel 100 as follows:



- Set *SIP Server IP Address* to **10.40.0.95**
- Set *User Name* to **test**
- Set *SIP Register Period* to **0**
- Set *SIP Trunk Profile* to **EdgeMarc SBC**
- Set *SIP Source Port* to **10060**
- Set *SIP Destination Port* to **5060**
- Check the *Enable Channel* checkbox

Click **OK**.

Figure 36: Edit channel 100

6. While on the *SIP Trunk Configuration* page, select channel 100 and click **Copy To**. Copy the configuration to channel 101.

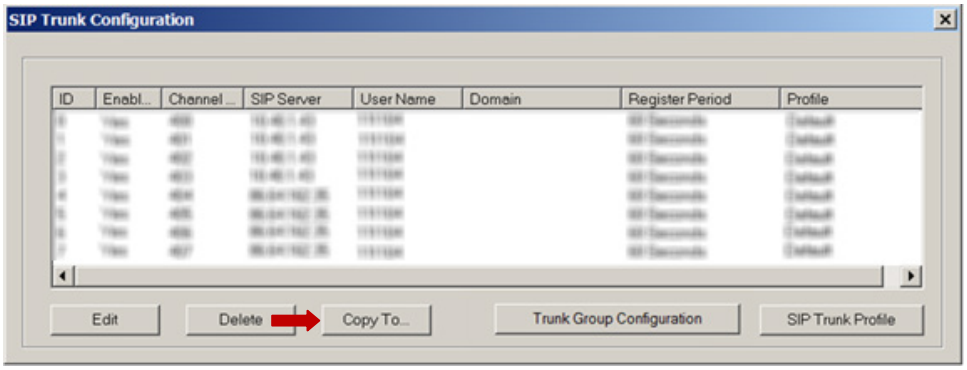


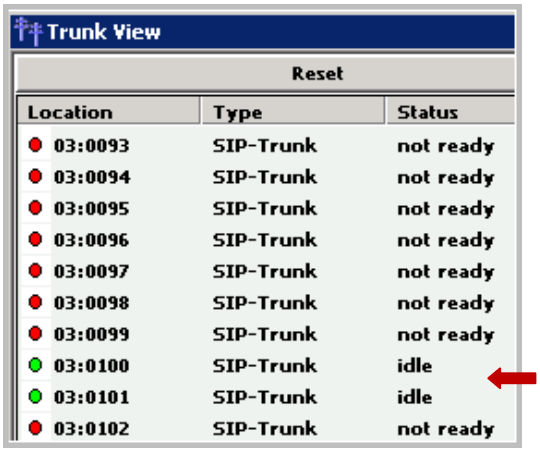
Figure 37: Use the "Copy To" button to copy channel 100 to channel 101

Note: The number of SIP Trunk channels must match the number of configured FXO ports on the AudioCodes gateway.

Verification

Now that you have configured MaxCS, verify that the system works correctly.

1. In MaxAdministrator, open Trunk view. Channel 3:0100 and 3:0101 should show a status of *Idle*.



Trunk View

Reset

Location	Type	Status
03:0093	SIP-Trunk	not ready
03:0094	SIP-Trunk	not ready
03:0095	SIP-Trunk	not ready
03:0096	SIP-Trunk	not ready
03:0097	SIP-Trunk	not ready
03:0098	SIP-Trunk	not ready
03:0099	SIP-Trunk	not ready
03:0100	SIP-Trunk	idle
03:0101	SIP-Trunk	idle
03:0102	SIP-Trunk	not ready

Figure 38: Channels 03:0100 and 03:0101 are idle

2. Make a PSTN call to 5102520000 from your mobile phone. The call should hit the trunk 03:0100 or 03:0101 with the DID number 162. Since DID number 162 is not configured, the call will be sent to the default IVR.

   For more information about in-call-routing with the DID number, refer to the MaxCS *Administration Manual* or press **F1** to open the Help system.

3. Making outbound PSTN calls through the SIP-Trunk is also possible. However, you will need to dial an extra-'9' to make a PSTN call. For example, if trunk access code '9' is assigned to the SIP-trunk 03:0100, then to call 15102520004, users must dial 9915102520004. You can set up out-call-routing in MaxAdministrator to insert an additional '9' automatically; refer to the MaxCS *Administration Manual* for instructions.

Move T.38 FAX ATA/Gateway Behind the EdgeMarc SBC

If you already have an ATA device dedicated for fax with a public IP address, you can either leave the device as-is, or you can move the ATA device behind the EdgeMarc SBC to save a public IP address.

The next figure depicts the configuration of a fax ATA behind an EdgeMarc SBC. Following are guidelines:

- The fax ATA device must be dedicated.
- Do not mix MaxCS FXS extensions and FXO ports with T.38 in the same AudioCodes gateway, because doing so might create SIP port conflicts. Refer to the *MP202 T.38 FAX Configuration Guide* for instructions.

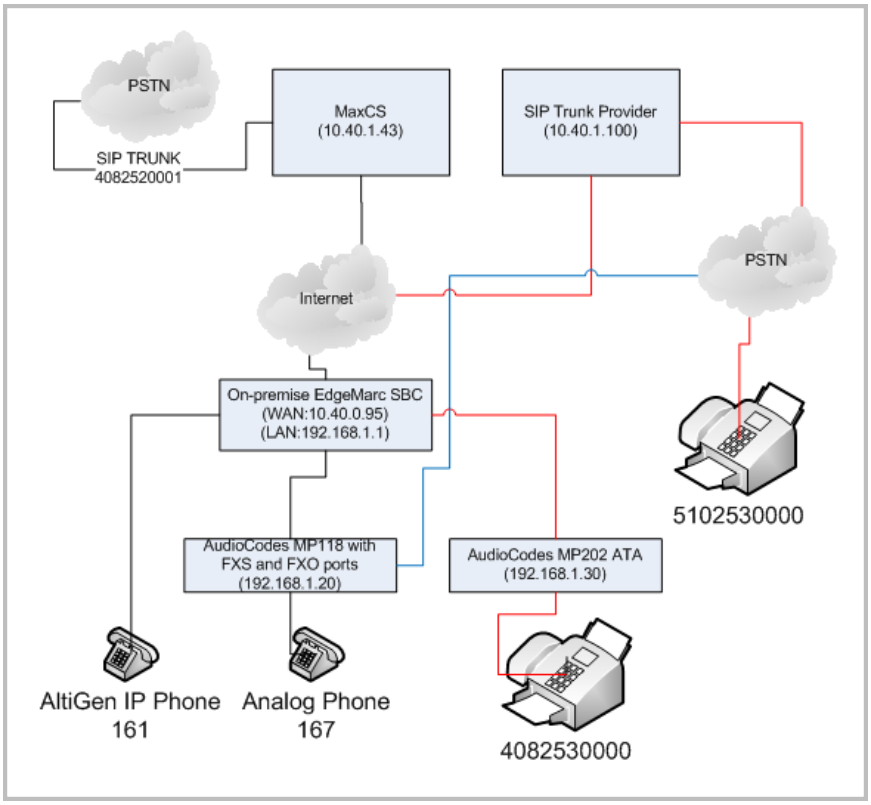


Figure 39: Diagram of Fax ATA device behind EdgeMarc SBC

To move FAX ATA/Gateway behind EdgeMarc SBC:

1. Consult the ATA/Gateway's user manual to change the fax ATA's public IP address to a private static IP address behind the EdgeMarc SBC. IP address 192.168.1.30 is used in this example.
2. If the SIP Trunk provider does not support SIP registration from an ATA device and the EdgeMarc's public IP address is different from the fax ATA's previous public IP address, then you will need to talk to your SIP trunk service provider to forward 4082530000 to the EdgeMarc SBC's public IP address, 10.40.0.95 in this example.

3. In the EdgeMarc configuration tool, select **VoIP** > **SIP**. In the Allowed SIP Proxies section, add your SIP service provider's IP address, (10.40.1.100 in our example).

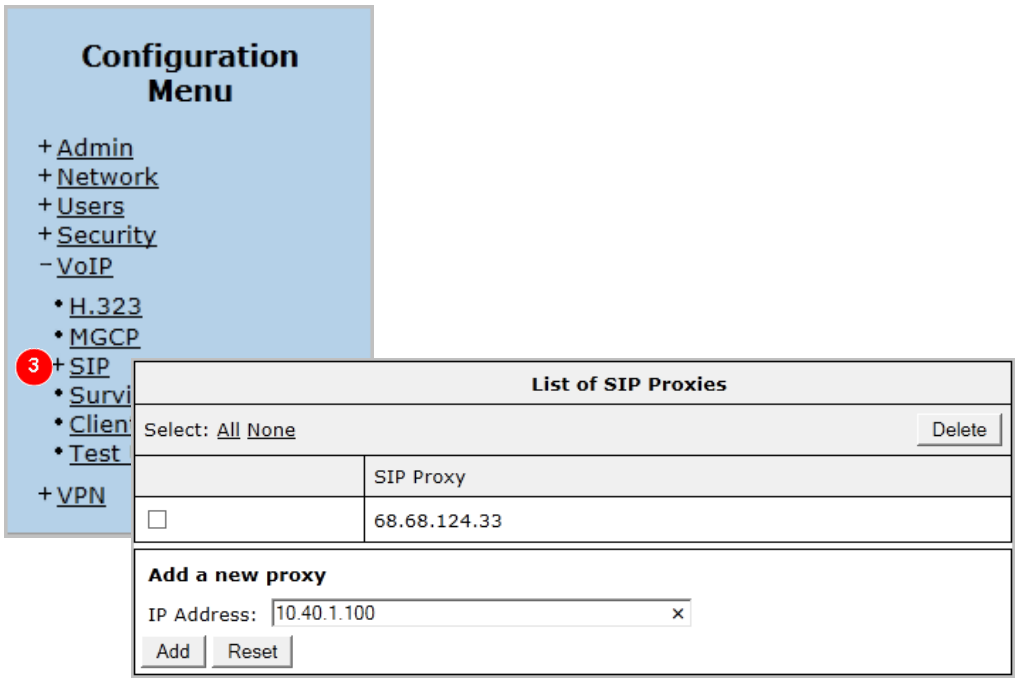
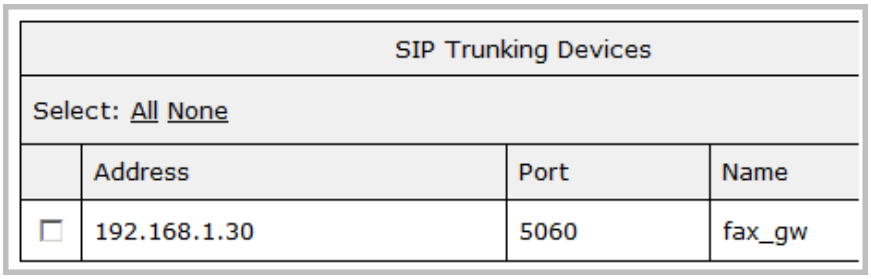


Figure 40: Add a new SIP proxy with your SIP service provider's IP address

4. Scroll to the bottom of the page and click **Submit**. Then click **Apply later**.
5. (If your SIP service provider supports SIP registration, skip this step.)

   Select **VoIP** > **SIP** > **ALG**. Add a new trunking device as follows:

   - Set *Action* to **Add new trunking device**
   - For *Name* enter **fax_gw**
   - For *Address* enter **192.168.1.30**
   - For *Port* enter **5060** (the MP202's SIP port number)



SIP Trunking Devices

Select: All None

	Address	Port	Name
☐	192.168.1.30	5060	fax_gw

Figure 41: Add a new SIP trunking device

6. (If your SIP service provider supports SIP registration, skip this step.)

   On the same page, add a new rule as follows:

   - Set *Action* to **Add new rule**.
   - Set *Type* to **Inbound**.
   - Set *Pattern-match* to **+14082530000**. The inbound fax call SIP INVITE from the header – this may vary from one service provider to another, some may not require the plus sign (+) or the 1. If your entry does not work, we recommend that you use Wireshark to capture the SIP trace to determine the correct prefix for your specific service provider.
   - Set *Trunking Device* to **fax_gw (192.168.1.30:5060)**

Verification

To confirm that your configuration works correctly, follow these steps.

1. Make outbound and inbound fax calls.
2. Unplug the EdgeMarc SBC's uplink and make an outbound PSTN fax call. The call should go through the AudioCodes gateway's FXO port.

Limitations

Known limitations:

- The T.38 fax line does not support survivability mode.
- The #81 and 82 hands-free modes and dial tone disabled mode are not supported.

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