

MAXCS Release 7.0

Application Note:  
Remote Survivability  
with AudioCodes Gateway

**Intended audience:**  
**AltiGen Authorized Partners**

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## About This Guide

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 AudioCodes gateway's FXO ports.

The examples in this guide are based upon an AudioCodes MP-118 gateway with 4 FXS and 4 FXO ports. Refer to the documentation for your gateway as needed (to find your AudioCodes user manual, searching for "LTRT-65417 MP-11x and MP-124 SIP User's Manual Ver 6.6.pdf" in your web browser).

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, on page 17.

- usa\_tones\_13\_NoHold.ini
- usa\_tones\_13\_NoHold.dat

**Note:** AltiGen does not provide general configuration support for AudioCodes products.

A support agreement with AudioCodes is required.

## Related Documents

- For instructions on configuring analog extensions using an AudioCodes MP-118 gateway, refer to *Application Note: Configuring Analog Extensions with AudioCodes Gateways*. This guide is available from your AltiGen representative.

**IMPORTANT!** If you plan to have your MP-118 gateway support both FXS and FXO, we recommend that you to configure your analog extensions first, and then return to this guide to set up survival mode. There are specific steps for configuring analog extensions that are not covered in this guide, including Proxy Group setup.

- For instructions on configuring Polycom IP phones, refer to the *Polycom IP Phone Configuration Guide*.

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## 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 via an AudioCodes gateway's FXO ports. Inbound PSTN calls to the local PSTN trunk can still be routed to a local extension.

This configuration supports up to 25 local (IP) phones in on-premise site or remote branch office, where the maximum 25 local phones is limited by the gateway's Stand Alone Survivability (SAS). If you have more than 25 local IP phones, or if you want to add traffic shaping/QOS, refer the AltiGen document, *Application Note: Remote Survivability with EdgeMarc SBC and AudioCodes MP-118.* That guide explains how to set up the EdgeMarc SBC and AudioCodes. If you decide to use an EdgeMarc SBC, skip this document (the configurations may differ).

If you already have configured your AudioCodes MP-118 gateway for analog extensions, you can use the same gateway to configure the survival FXO ports. You also can configure a separate MP-118 gateway with FXO ports.

## Architecture

Figure 1 shows the survival setup with an AudioCodes MP-118.

While the AltiGen phone (extension 161), the Polycom IP phone (extension 162), and the analog phone behind an MP-118 (extension 167) can reach MaxCS (10.40.1.43), the phone users of those extensions can make calls, receive calls, and use all of the MaxCS call features.

While phones can reach MaxCS, incoming calls should come from the SIP trunk (4082520001). However, if a PSTN user (5102520004) dials the survival PSTN trunk 5102520000, the MP-118 gateway can only send this call to the pre-defined Polycom extension (162) or the extension behind MP-118 (167) in this example. If the pre-defined extension number is an AltiGen phone (161), the caller will hear a busy signal, because the AltiGen IP phone is currently registered to MaxCS (10.40.1.43). (Polycom phones can register with two SIP proxies concurrently.)

While extensions 161, 162 and 167 cannot reach MaxCS, PSTN outbound calls from those extensions will go through the MP-118's FXO ports. These extensions can still call each other, because they all fall back to registering on the AudioCodes M-P1x18s SAS SIP port. However, those users will lose most of the features provided by MaxCS during that time, including voicemail, transfers, and conferences.

Incoming calls to 5102520001 will be routed to a pre-configured extension 162 in this example.

When extensions 161, 162 and 167 can reach MaxCS again, their users will have access to all of the MaxCS features again within 2 minutes.

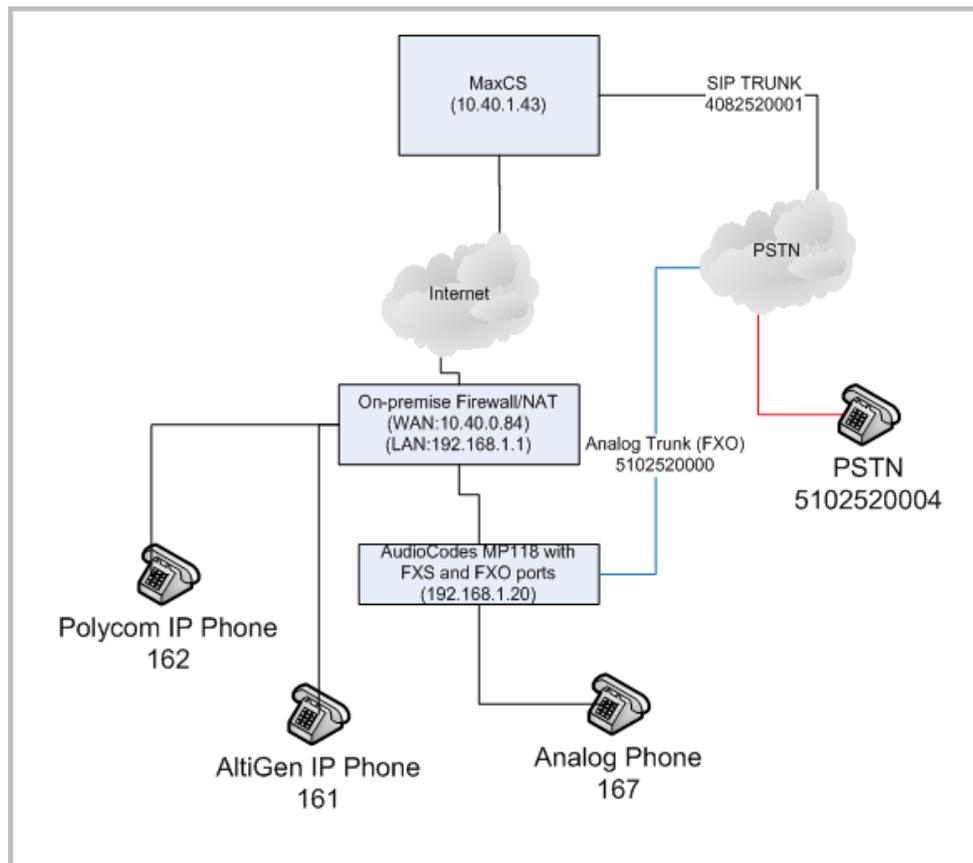


Figure 1: Survival configuration with AudioCodes MP-118

## Prerequisites

To set up survivability using an AudioCodes MPxxx gateway, your environment must meet the following requirements:

- You must be running either MaxCS Release 7.0 Enterprise or MaxCS Private Cloud.
- Proper configuration for MaxCS behind NAT is required (including the port forwarding and Enterprise Manager settings). Any AltiGen IP phones behind NAT should already be working correctly behind the on-premise firewall.
- Your AltiGen IP phones must be running on Firmware 2xA3 or later.
- If you have any Polycom IP phones, they must have been configured correctly. Refer to the AltiGen document *Polycom IP Phone Configuration Guide* for instructions. This guide is available from your AltiGen representative
- You must have an AudioCodes MP-118 gateway with FXO ports available.
- You must add the Firewall/NAT WAN address (in Figure 1, this is 10.40.0.84) to the MaxCS *Trusted SIP Device List*. If this IP address is not included in the list, then it will be treated as a malicious SIP device due to excessive SIP messages from that address.
- Make sure that your AudioCodes gateway is using the correct firmware version. To determine this, click the **Home** button and make sure the device has version 6.60A.265.010.

## AudioCodes Configuration

If you already have an AudioCodes MP-118 gateway that you have configured for analog extensions, you may have already performed some of the steps in this guide.

If you plan to have your gateway support both FXS and FXO, we recommend that you to configure your analog extensions first, and then return to this guide to set up survival mode.

**Note:** If you have analog phones currently logged into MaxCS using FXS ports behind the same gateway, after you configure that gateway for survival support, **those analog phones will stop working**. You will need to perform a few additional steps (which are described in this guide) to make both the FXO and FXS ports work correctly.

**Important!** While working in the AudioCodes configuration application, make sure that **Full** is always selected for the *Configuration* menu, so that you can see all of the menu options.

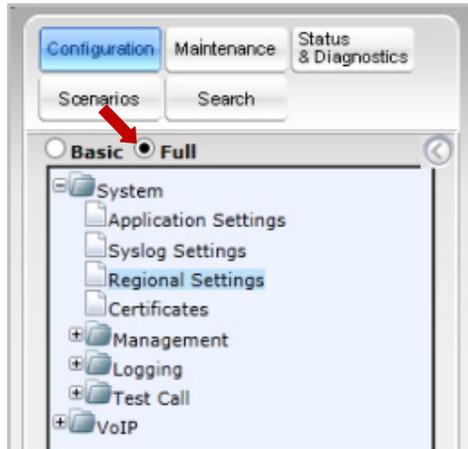


Figure 2: Choose Full for the AudioCodes Configuration menu

Also, make sure you click **Submit** after each change, and later click **Burn**, or your configuration changes will be lost after the gateway is restarted.

1. Select **VoIP > Network > IP Interface 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.)

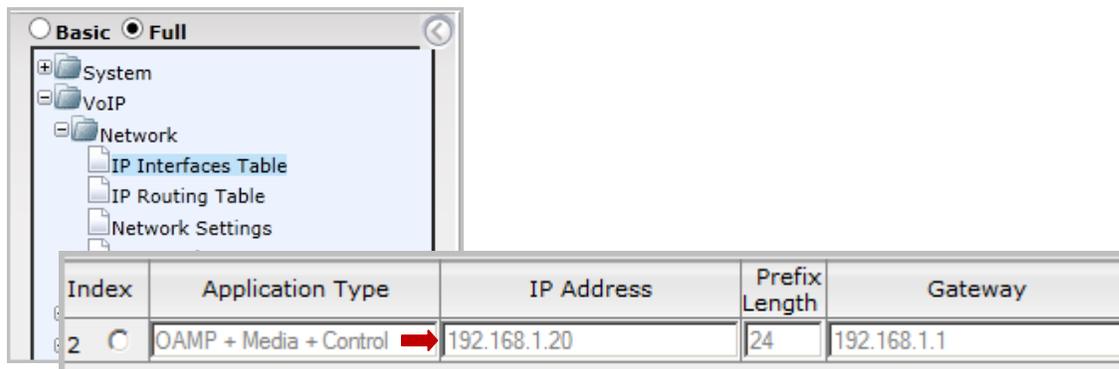
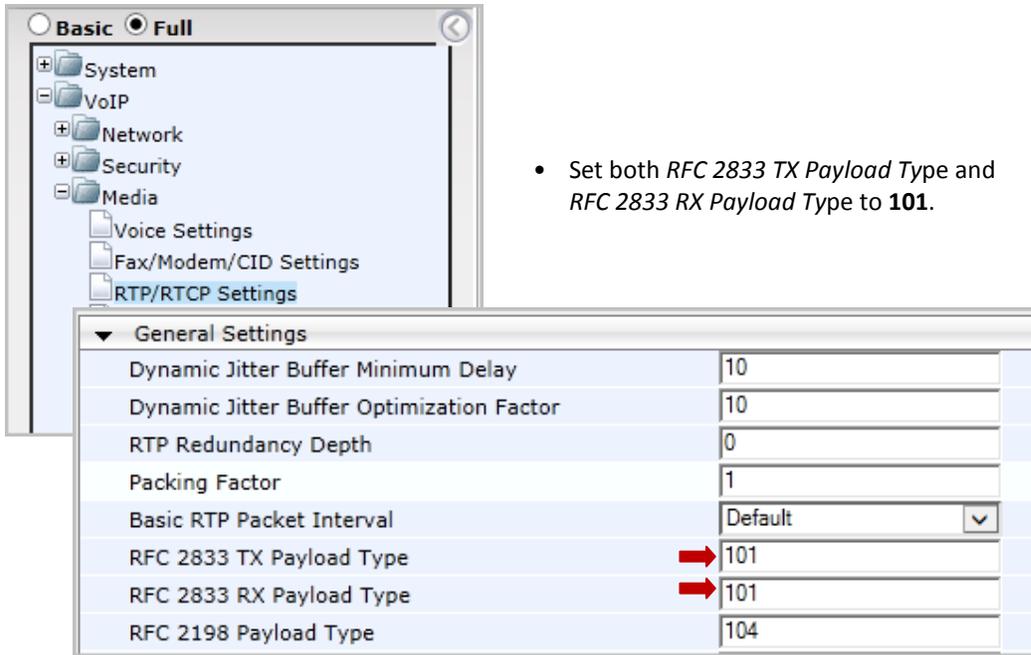


Figure 3: Check the Network IP settings

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



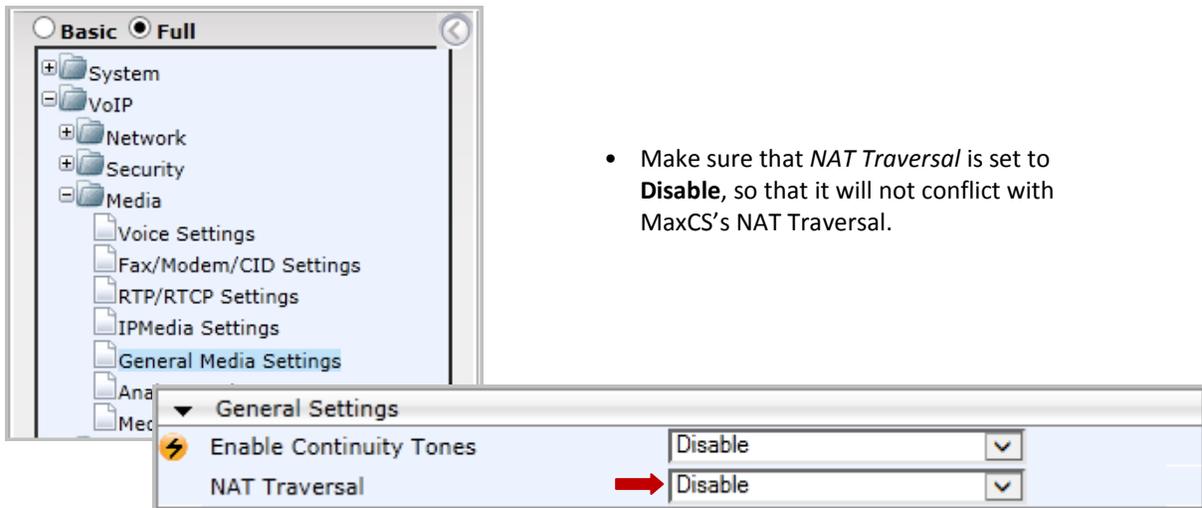
The screenshot shows the configuration window for RTP/RTCP Settings. The left sidebar shows a tree view with 'Media' expanded and 'RTP/RTCP Settings' selected. The main area shows the 'General Settings' tab with the following values:

Dynamic Jitter Buffer Minimum Delay	10
Dynamic Jitter Buffer Optimization Factor	10
RTP Redundancy Depth	0
Packing Factor	1
Basic RTP Packet Interval	Default
RFC 2833 TX Payload Type	101
RFC 2833 RX Payload Type	101
RFC 2198 Payload Type	104

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

Figure 4: Configure the Media RFC 2833 Payload settings

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



The screenshot shows the configuration window for General Media Settings. The left sidebar shows a tree view with 'Media' expanded and 'General Media Settings' selected. The main area shows the 'General Settings' tab with the following values:

Enable Continuity Tones	Disable
NAT Traversal	Disable

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

Figure 5: Configure the general media settings

4. Select **VoIP > Applications Enabling**.

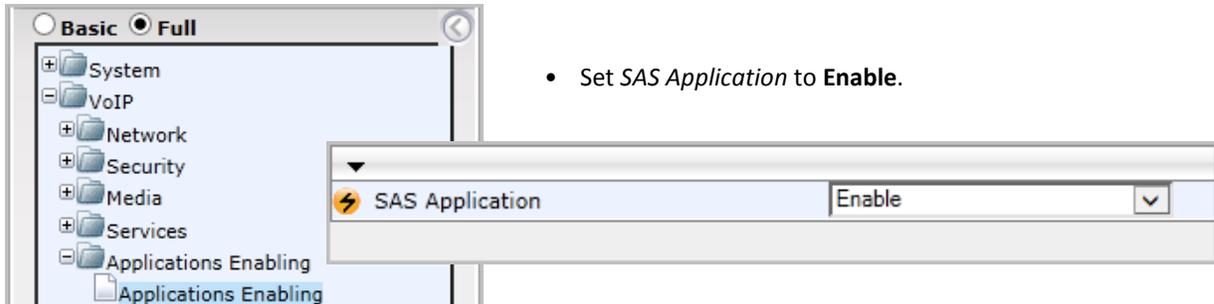


Figure 6: Enable the SAS Application option

5. Submit this change. On the toolbar, click **Burn** to save this configuration.



Figure 7: Click the Burn button to save the configuration

6. Reboot the AudioCodes gateway.

7. Make sure you select **Full** for the Configuration menu, as you did on page 5. Select **Control Network > Proxy Sets Table >**. Configure *Proxy Set ID 0* as follows:

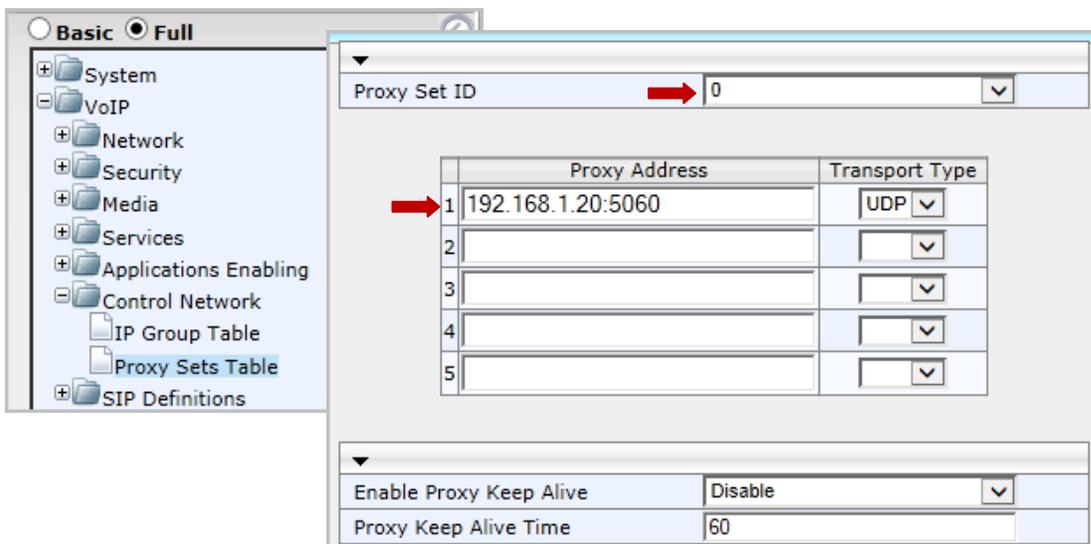


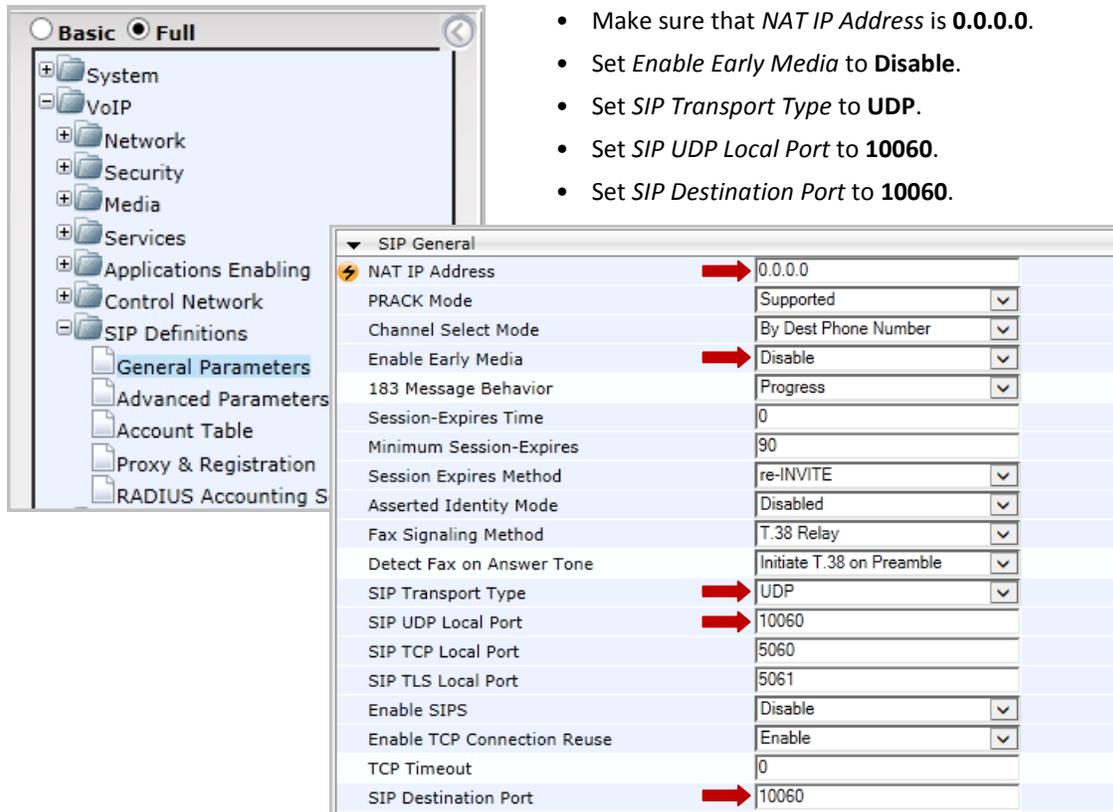
Figure 8: Configure proxy set 0 with the IP Address of the MP-118

- Set *Proxy Address 1* to **192.168.1.20:5060**. This is MP-118 SAS IP address and port, which you will set up shortly. When there is an incoming call to the gateway's FXO port, the call will loop back to

192.168.1.20:5060 with a predefined extension number, (162 in this example). This predefined number is configured in **VoIP>GW and IP to IP > Analog Gateway > Automatic Dialing**.

- Set the *Transport Type* to **UDP**.
- Set *Enable Proxy Keep Alive* to **Disable**.

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

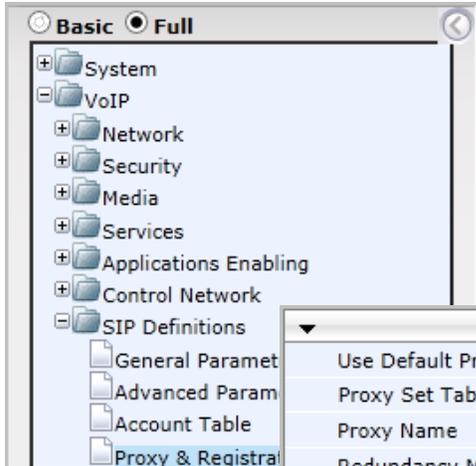


SIP General	
NAT IP Address	0.0.0.0
PRACK Mode	Supported
Channel Select Mode	By Dest Phone Number
Enable Early Media	Disable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	UDP
SIP UDP Local Port	10060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPS	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	0
SIP Destination Port	10060

- 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 9: Configure the general SIP parameters

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



- Set *Use Default Proxy* to **Yes**.
- Set *Enable Registration* to **Enable**.
- Set *Registrar IP Address* to **192.168.1.20**. (The MP-118's SAS SIP address.)
- Set *Registrar Transport Type* to **UDP**.
- Set *Registration Time* to **160**.

Use Default Proxy	Yes
Proxy Set Table	
Proxy Name	
Redundancy Mode	Homing
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Standard Mode
Enable Registration	Enable
Registrar Name	
Registrar IP Address	192.168.1.20
Registrar Transport Type	UDP
Registration Time	160
Re-registration Timing [%]	50

Figure 10: Configure the proxy and registration settings

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

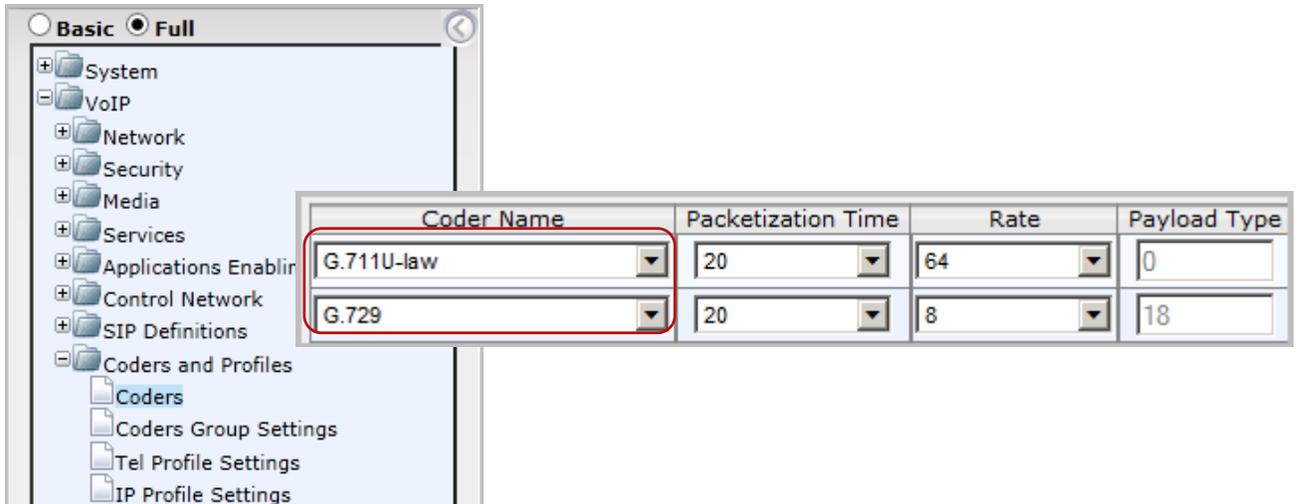
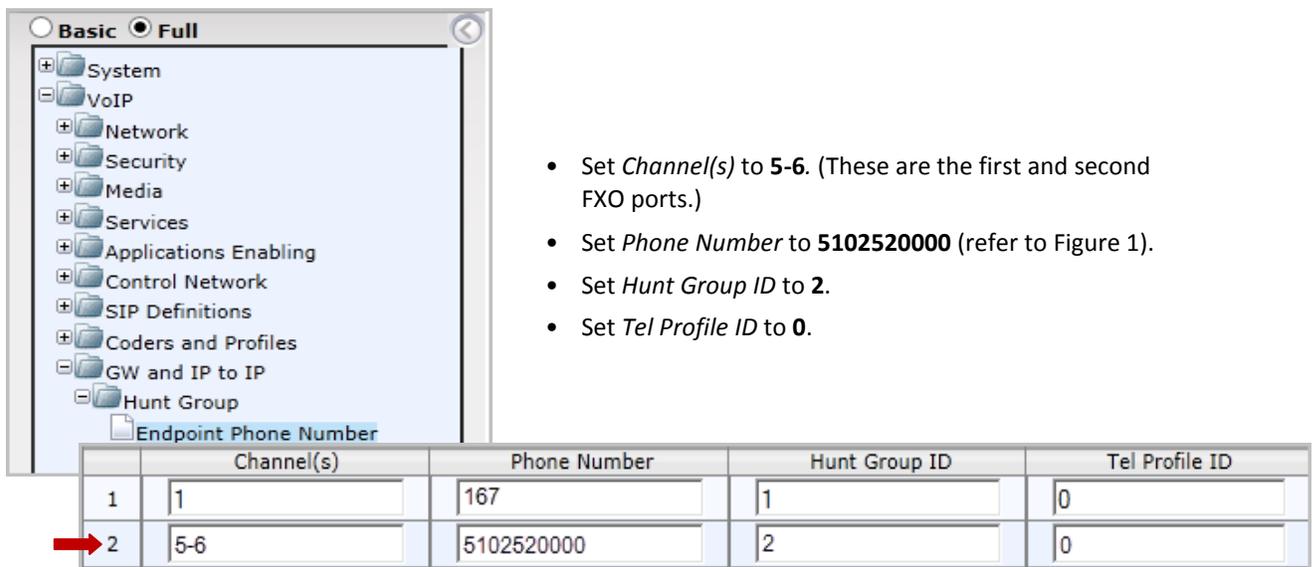


Figure 11: Confirm that G.711U-law and G.729 are listed in Coders and Profiles page

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



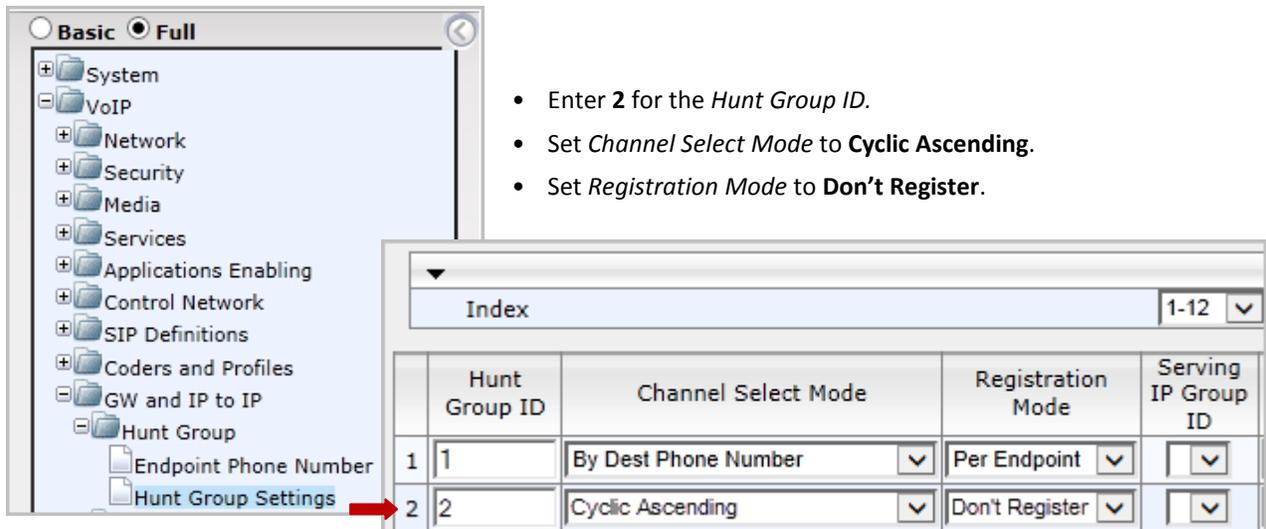
- Set *Channel(s)* to **5-6**. (These are the first and second FXO ports.)
- Set *Phone Number* to **5102520000** (refer to Figure 1).
- Set *Hunt Group ID* to **2**.
- Set *Tel Profile ID* to **0**.

Figure 12: Configure the endpoint phone extension

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

Do not configure an FXO channel without attaching a physical PSTN trunk into the FXO port. In other words, if you configure both channel 5 and channel 6, but you only attach a PSTN trunk to channel 5, then 50% of the outbound PSTN calls may fail.

12. 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):



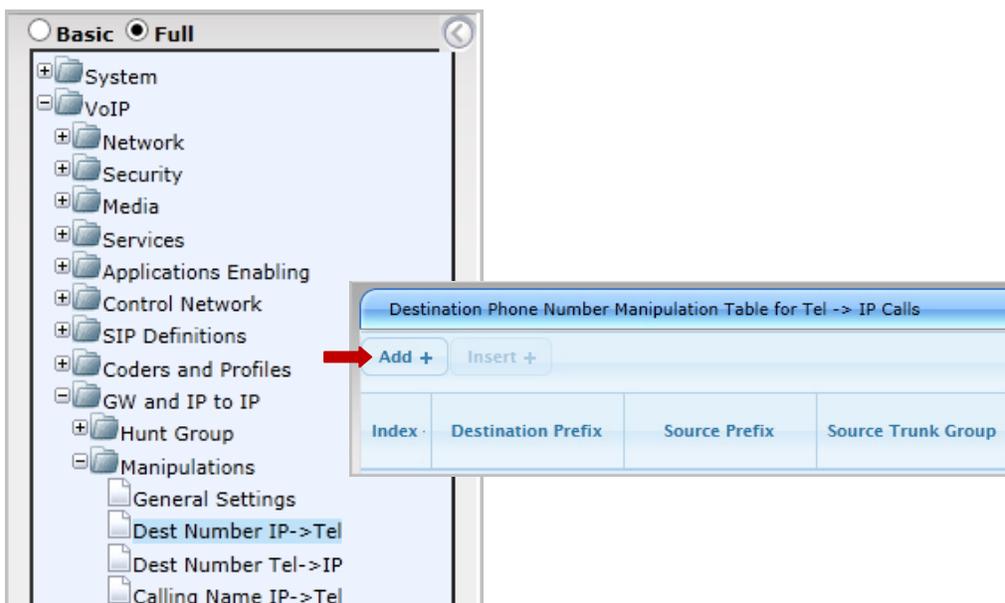
- Enter **2** for the *Hunt Group ID*.
- Set *Channel Select Mode* to **Cyclic Ascending**.
- Set *Registration Mode* to **Don't Register**.

Index	Hunt Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID
1	1	By Dest Phone Number	Per Endpoint	
2	2	Cyclic Ascending	Don't Register	

Figure 13: Configure the hunt group settings

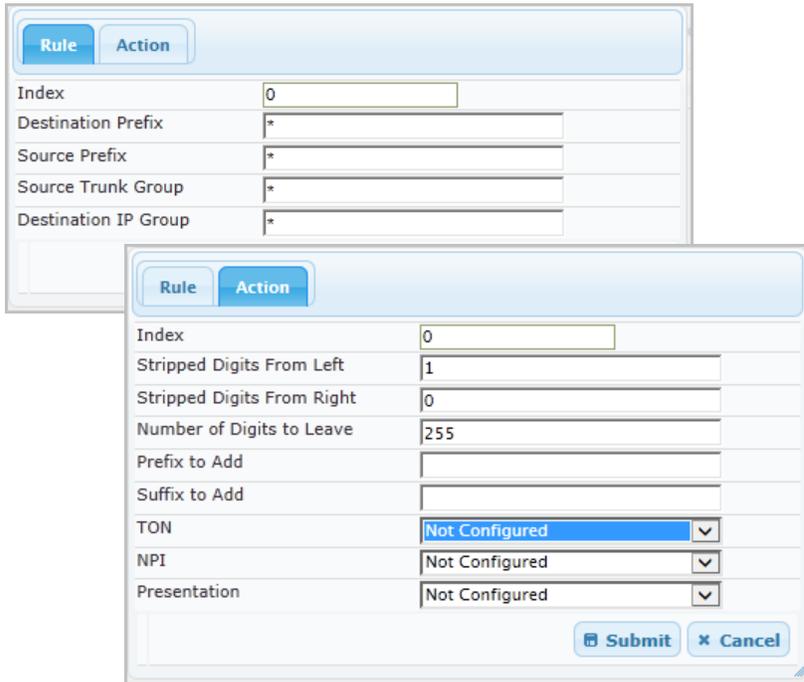
13. With MaxCS, users must dial a trunk access code (9) first in order to place a PSTN call. When MaxCS is in Survival mode, the MP-118 must strip out the preceding '9' before sending the number.

Select **VoIP > GW and IP to IP > Manipulations > Dest Number IP -> Tel**. Click the **Add +** button.



Index	Destination Prefix	Source Prefix	Source Trunk Group

Figure 14: Add a new manipulation rule



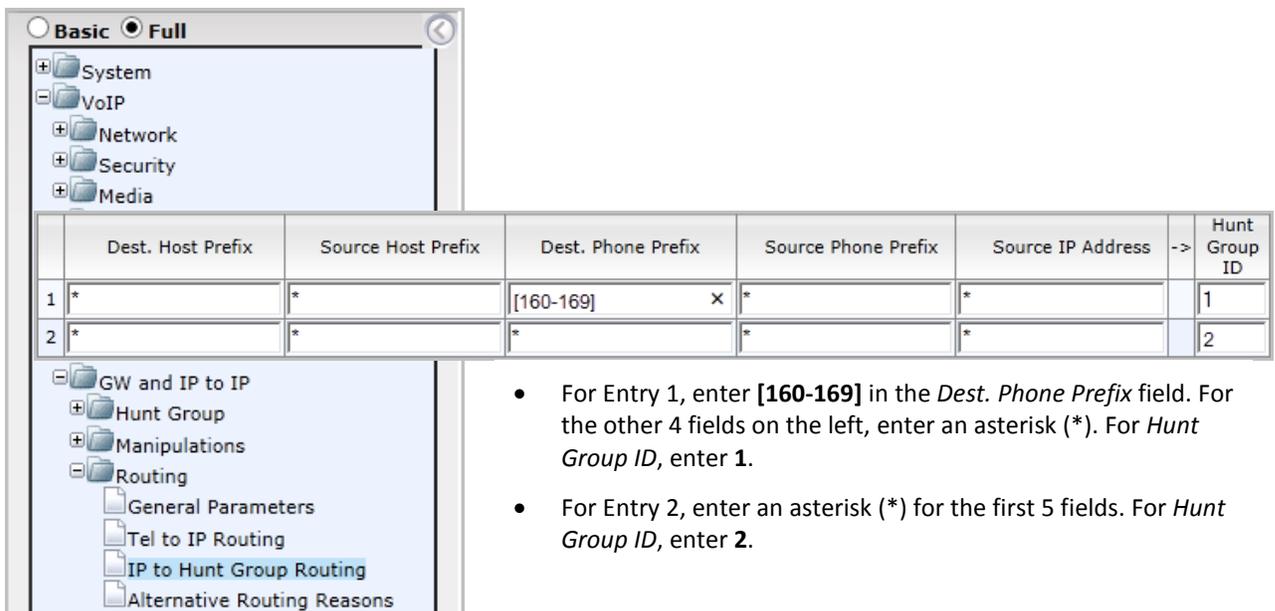
- In the *Rule* section, set *Index* to 0. For the remaining four options, enter an asterisk (\*).
- In the *Action* section, set *Stripped Digits From Left* to 1.

Figure 15: Strip one digit from the left

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

**Note:** If there are no analog extensions behind this MP-118 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.

If you have analog extension behind this gateway, the following settings are required:

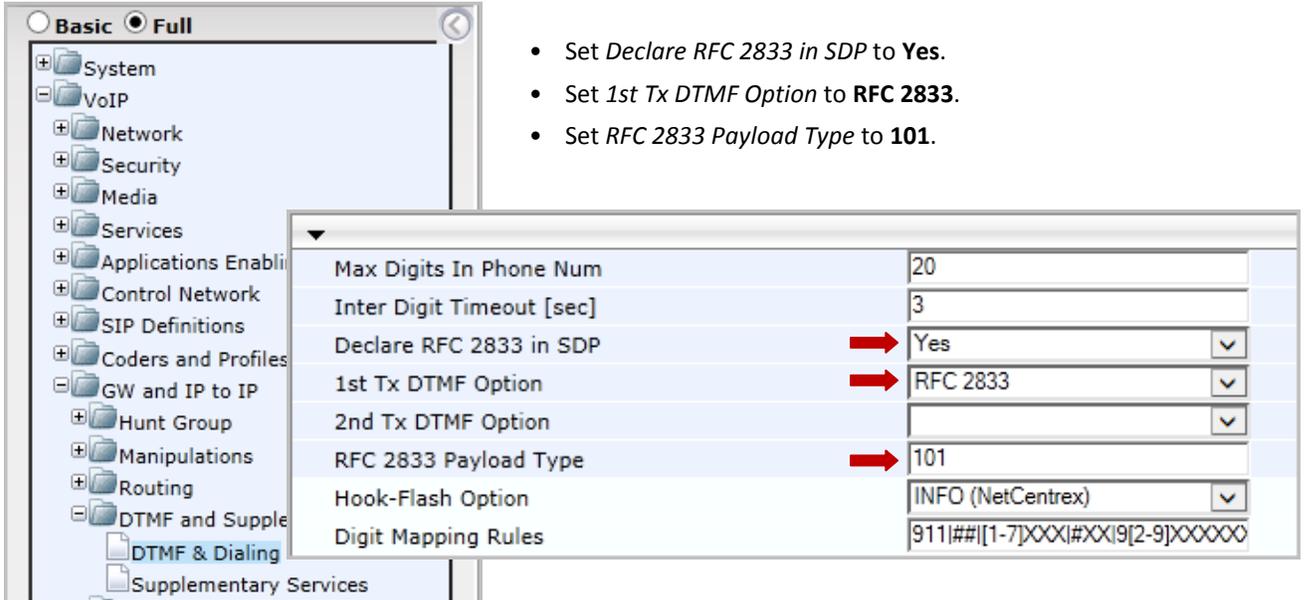


	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	-> Hunt Group ID
1	*	*	[160-169] ×	*	*	1
2	*	*	*	*	*	2

- 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 16: Configure hunt group routing rules

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

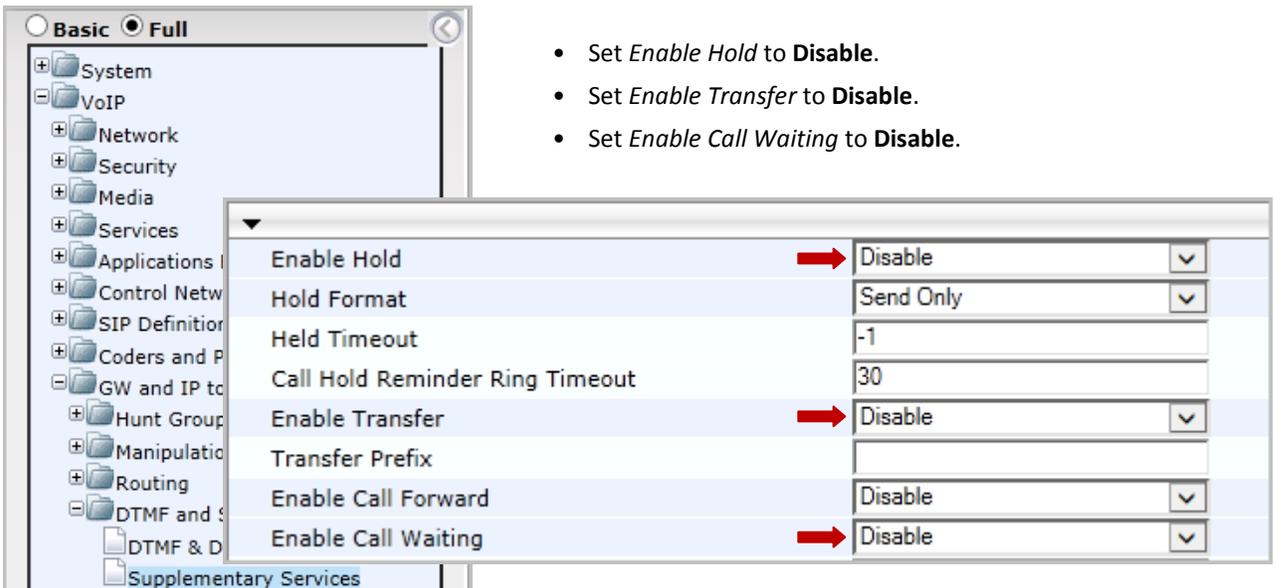


Max Digits In Phone Num	20
Inter Digit Timeout [sec]	3
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	
RFC 2833 Payload Type	101
Hook-Flash Option	INFO (NetCentrex)
Digit Mapping Rules	911 ## [1-7]XXX ##XX 9[2-9]XXXXXX

- Set *Declare RFC 2833 in SDP* to **Yes**.
- Set *1st Tx DTMF Option* to **RFC 2833**.
- Set *RFC 2833 Payload Type* to **101**.

Figure 17: Configure DTMF and dialing settings

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

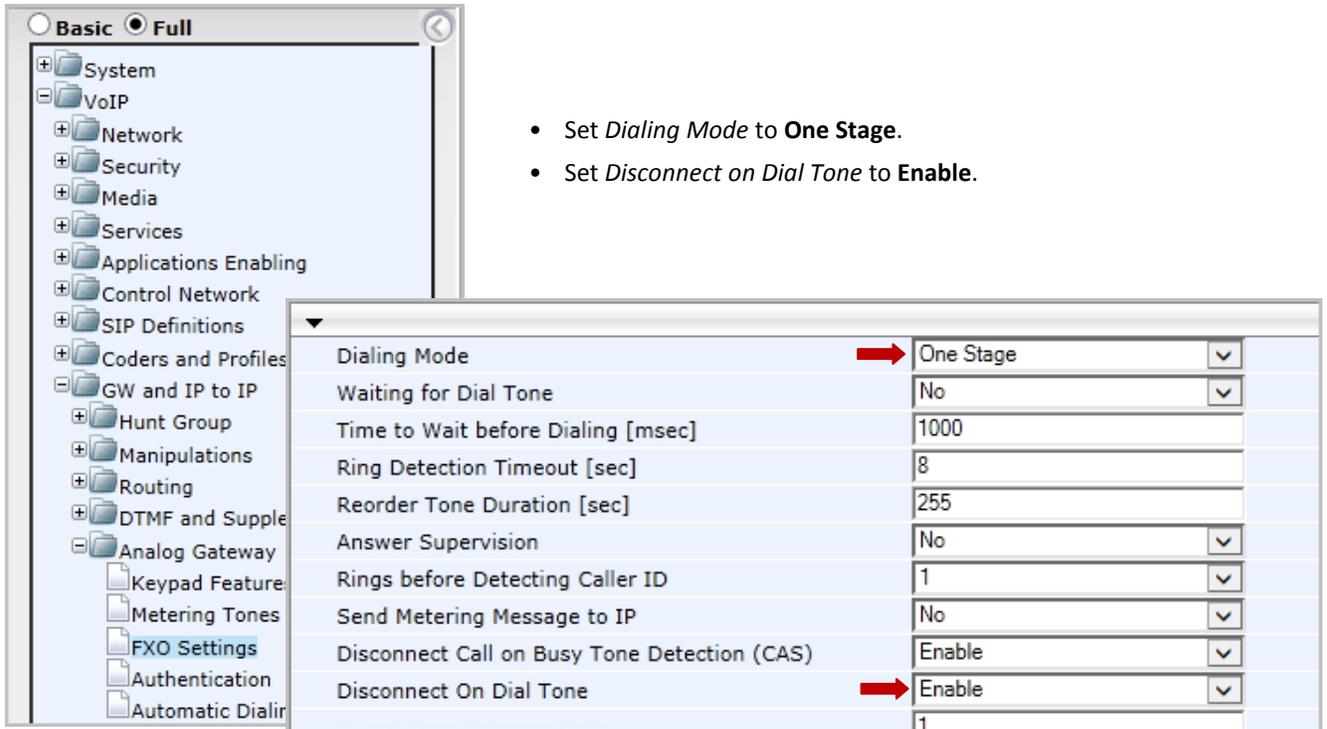


Enable Hold	Disable
Hold Format	Send Only
Held Timeout	-1
Call Hold Reminder Ring Timeout	30
Enable Transfer	Disable
Transfer Prefix	
Enable Call Forward	Disable
Enable Call Waiting	Disable

- Set *Enable Hold* to **Disable**.
- Set *Enable Transfer* to **Disable**.
- Set *Enable Call Waiting* to **Disable**.

Figure 18: Disable the Hold, Transfer, and Call Waiting features

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

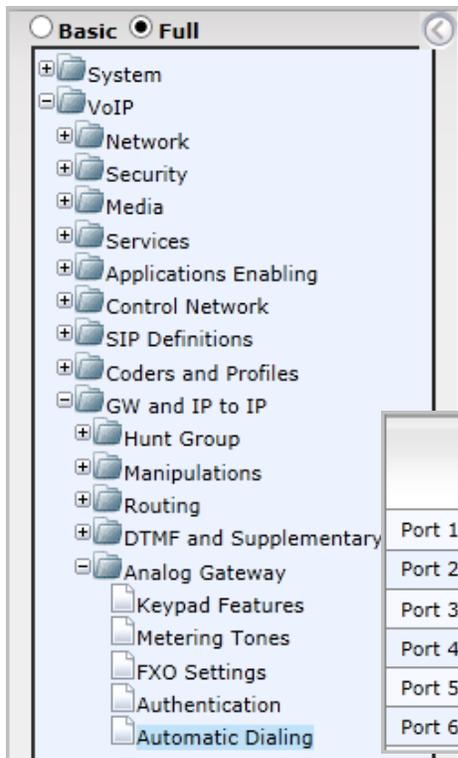


Dialing Mode	One Stage
Waiting for Dial Tone	No
Time to Wait before Dialing [msec]	1000
Ring Detection Timeout [sec]	8
Reorder Tone Duration [sec]	255
Answer Supervision	No
Rings before Detecting Caller ID	1
Send Metering Message to IP	No
Disconnect Call on Busy Tone Detection (CAS)	Enable
Disconnect On Dial Tone	Enable
	1

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

Figure 19: Configure FXO settings

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

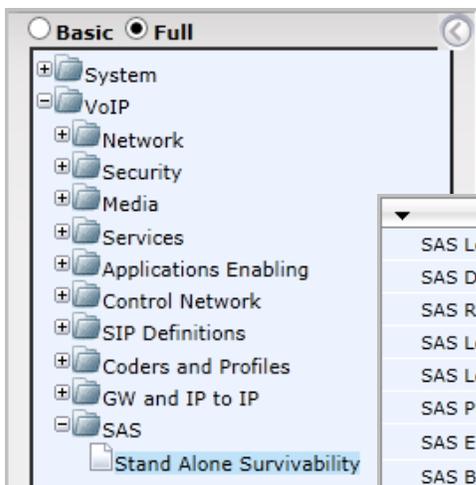


- Set both *Port 5 FXO* and *Port 6 FXO* to **162**. During Survival mode, incoming calls to FXO ports will be sent to extension 162.

Gateway Port	Destination Phone Number
Port 1 FXS	
Port 2 FXS	
Port 3 FXS	
Port 4 FXS	
Port 5 FXO	→ 162
Port 6 FXO	→ 162

Figure 20: Route incoming calls to FXO ports to extension 162

19. Select **VoIP > SAS > Stand Alone Survivability**.



- Set *SAS Local SIP UDP Port* to **5060**.
- Set *SAS Binding Mode* to **1-User Part Only**.
- Set *SAS Survivability Mode* to **Always Emergency**.

SAS Local SIP UDP Port	→ 5060
SAS Default Gateway IP	
SAS Registration Time	20
SAS Local SIP TCP Port	5080
SAS Local SIP TLS Port	5081
SAS Proxy Set	0
SAS Emergency Numbers	
SAS Binding Mode	→ 1-User Part Only
SAS Survivability Mode	→ Always Emergency

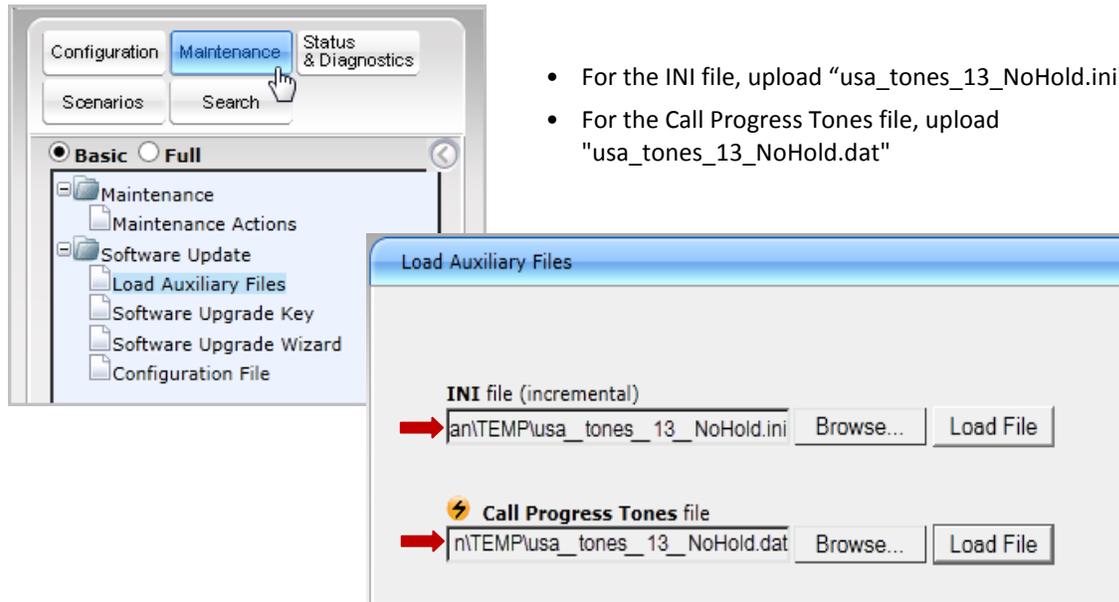
Figure 21: Configure stand-alone survivability settings

20. 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

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

Click the **Maintenance** button above the menu (see the next figure). 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 22: Upload the .INI and .DAT files

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

21. Submit this last change. On the toolbar, click **Burn**. Proceed to the next section.

## AltiGen IP Phone Configuration

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 phones will need a few extra steps to implement survivability.

1. On the AltiGen IP phone, press the **Menu** button.
2. In the menus, navigate to **System > Emergency Gateway**.
3. Enter the AudioCodes gateway's IP address (in our example, this is 192.168.1.20).
4. In the 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 entry, change the value to **No**.

## Polycom IP Phone Configuration

To configure Polycom phones to work with MaxCS 7.0, refer to the document, *Polycom IP Phone Configuration Guide*. However, to support MP-118 SAS with Polycom phones, additional configuration steps are required.

In that guide, refer to the section *Create the Extension-Specific Configuration Files*. A new configuration file, "162.cfg," should have the following content (the new statements are shown in red text):

```
<?xml version="1.0" encoding="UTF-8" standalone="yes"?>
<!-- Extension Registration Configuration File -->
<phone1>
<reg reg.1.displayName="D162" reg.1.address="162" reg.1.label="162" reg.1.auth.userId="162"
reg.1.auth.password="5656" reg.1.server.1.address="10.40.1.43" reg.1.server.1.expires="90" />
<msg msg.bypassInstantMessage="1" msg.mwi.1.callBackMode="contact"
msg.mwi.1.callBack="###@10.40.1.43" msg.mwi.1.subscribe="162@10.40.1.43" />
<AudioCodesSAS
  voIpProt.SIP.outboundProxy.port=""
  reg.1.server.2.port="5060"
  reg.1.server.2.address="192.168.1.20"
  reg.1.server.2.expires="90"
/>
</phone1>
```

1. On the FTP server, update the extension configuration file, "162.cfg."
2. Reboot the Polycom phone that is extension 162.

The Polycom phone will register to both 10.40.1.43:10060 (MaxCS) and to 192.168.1.20:5060 (MP-118 SAS). When 10.40.1.43 is not reachable, then outbound calls will be routed to 192.168.1.20:5060 (the MP-118 SAS gateway).

## AudioCodes Configuration for Both FXS and FXO

This section describes how to configure both FXS and FXO support on the same MP-118 gateway.

If you have analog phones that are currently logged into MaxCS using FXS ports on the same gateway, after you configure the MP-118 gateway for survivability support **those analog phones will not work**.

This section explains the additional steps you must take to make both the FXO and FXS ports on a single gateway work correctly.

Some guidelines:

- If you have implemented survivability mode, then later you perform the steps in *Configuring Analog Extensions with AudioCodes Gateways* to configure FXS support, you will need to go through the steps in this document, again, to correct the configuration. Then follow the additional configurations steps listed here.
- If you have a new MP-118 gateway *in addition* to the MP-118 gateway that you configured for survivability, refer to *Configuring Analog Extensions with AudioCodes Gateways* to set up analog support on the new gateway. If, later, you want to configure survivability on the same gateway that is configured for analog support, follow the configurations steps listed here.

**Note:** In configurations where there are multiple MP-118 gateways, the survival FXO ports can only exist on **one** of the MP-118 gateways. In other words, the FXO ports on any new MP-118 gateways will be wasted.

1. On the MaxCS system, run regedit.exe.
2. Navigate to one of these subkeys:  
 On a 32-bit Windows system:  
 HKEY\_LOCAL\_MACHINE \ SOFTWARE \ AltiGen Communications, Inc. \ AltiWare \ Service Providers \ SIPSP  
 On a 64-bit Windows system:  
 HKEY\_LOCAL\_MACHINE \ SOFTWARE \ Wow6432Node \ AltiGen Communications, Inc. \ AltiWare \ Service Providers \ SIPSP
3. Check the entry EnableSIPOptionBehindNAT. If the value is not 1, then change the value to 1. Save the change.
4. Reboot the MaxCS system. (If the value is already 1, you do not need to reboot the system.)
5. After the system is rebooted, if you have SIP trunks connected to the MaxCS system, make sure the SIP trunks are still working.

	ConfigDllPath	REG_SZ	C:\AltiServ\SP\SIPS
	DllPath	REG_SZ	C:\AltiServ\SP\SIPS
	EnableSIPOptionBehindNAT	REG_DWORD	0x00000001 (1)
	MaxBackFiles	REG_DWORD	0x00000032 (50)
	SipDIDExt	REG_DWORD	0x00000001 (1)

Figure 23: The registry entry EnableSIPOptionBehindNAT

6. In AudioCodes, under *Configuration* choose **VoIP > Control Network > IP Group Table**. Click the **Add +** button.

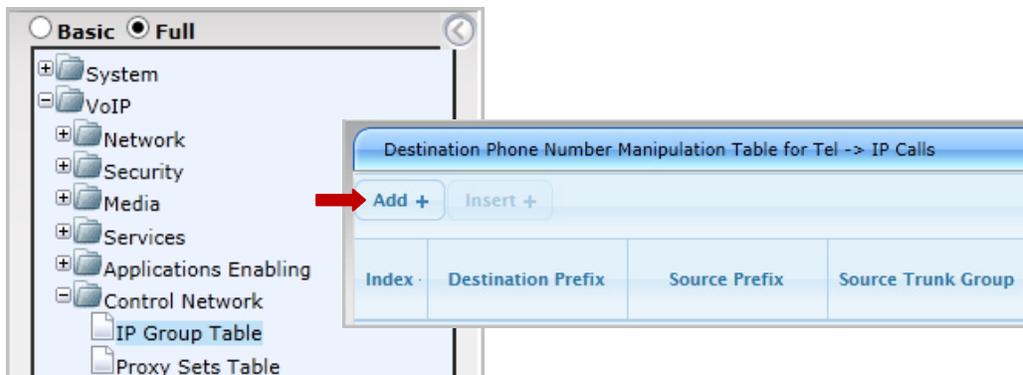


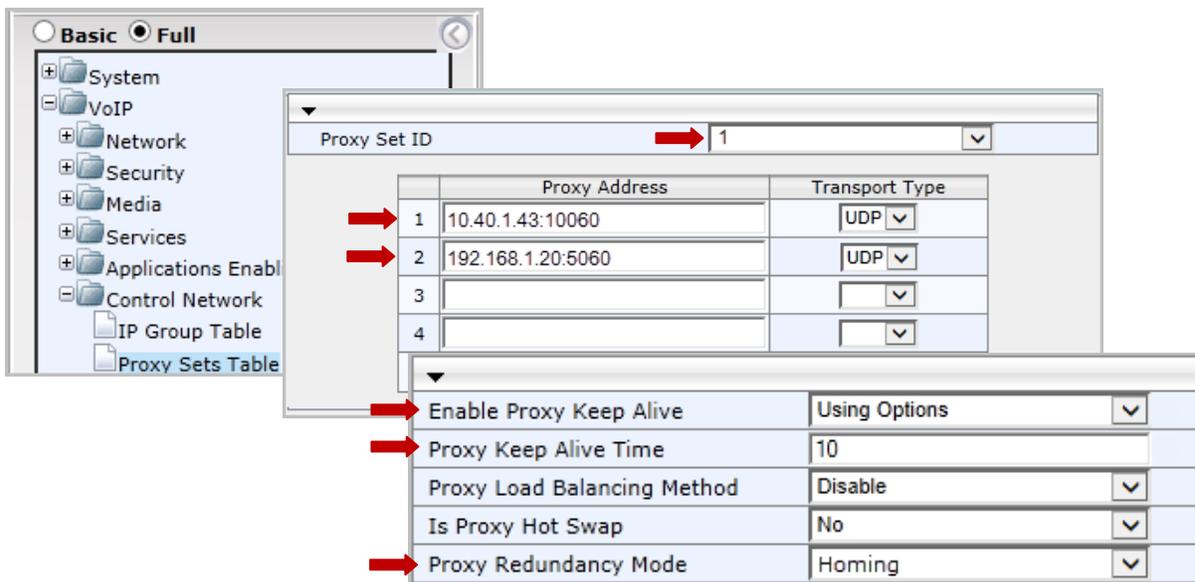
Figure 24: Add a new manipulation rule

7. Add a new entry as follows:

Common Gateway	
Index	1
Description	FXS Grp
Proxy Set ID	1
SIP Group Name	FXSSIP
Contact User	
Local Host Name	
IP Profile ID	1 <span style="float: right;">x</span>

- Set *Index* to **1**.
- For *Description* enter **FXS Grp**.
- For *Proxy Set ID* enter **1**.
- For *SIP Group Name* enter **FXSSIP**.
- For *IP Profile ID* enter **1**.

Figure 25: Configure the new rule



Proxy Set ID		
Proxy Set ID	1	
Index	Proxy Address	Transport Type
1	10.40.1.43:10060	UDP
2	192.168.1.20:5060	UDP
3		
4		
Enable Proxy Keep Alive	Using Options	
Proxy Keep Alive Time	10	
Proxy Load Balancing Method	Disable	
Is Proxy Hot Swap	No	
Proxy Redundancy Mode	Homing	

Figure 26: Configure Proxy Set ID 1

- For Entry 1, set the *Proxy Address* to the MaxCS SIP IP address and port (in our example, this is **10.40.1.43:10060**). Set the *Transport Type* to **UDP**.
- For Entry 2, set *Proxy Address* to the AudioCodes MP-118's SAS IP address and port (in our example, this is **192.168.1.20.5060**). Set the *Transport Type* to **UDP**.
- Set *Enable Proxy Keep Alive* to **Using Options**.
- Set *Proxy Keep Alive Time* to **10** (10 seconds).
- Set *Proxy Redundancy Mode* to **Homing**.

8. Select **VoIP > GW and IP to IP > Hunt Group > Endpoint Phone Number**.

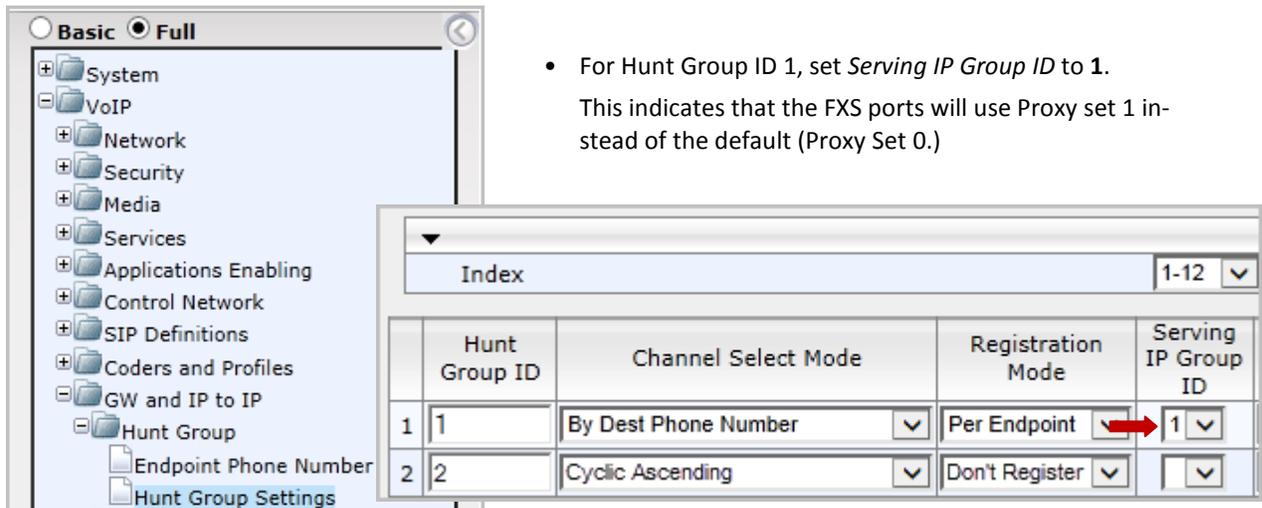
If you previously configured analog extensions, the Hunt Group ID for the FXS channels should be set to **1**.  
The Hunt Group ID for FXO channels should be set to **2**.

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	167	1	0
2	5-6	5102520000	2	0

Figure 27: Hunts groups for both analog extensions and survivability

If your Hunt Group ID values are not correct, refer to the steps in this guide and in the *Configuring Analog Extensions with AudioCodes Gateways* guide and make the appropriate corrections.

9. Select **VoIP > GW and IP to IP > Hunt Group > Hunt Group Settings**.



- For Hunt Group ID 1, set *Serving IP Group ID* to **1**.  
This indicates that the FXS ports will use Proxy set 1 instead of the default (Proxy Set 0.)

Figure 28: Configure Hunt Group 1 to use Serving IP Group 1

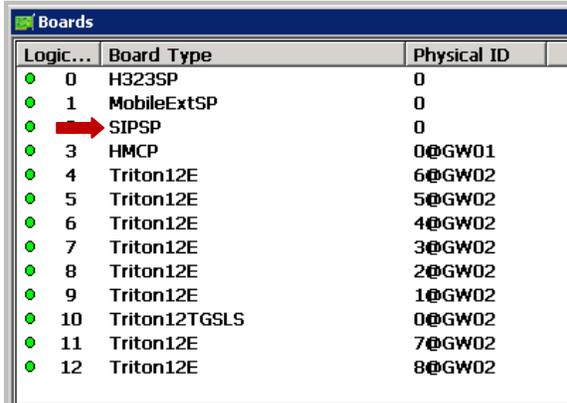
10. Submit this last change. On the toolbar, click **Burn** to save this configuration. Then proceed to the next section to verify that survivability is correctly configured.

## Verification

Now that you have configured the phones and the MP-118 gateway, verify that the settings are correct and that the phones work correctly.

Looking back at Figure 1, the AltiGen IP phone 161, the Polycom IP phone 162 and the analog phone 167 behind the gateway should all be able to make calls, receive calls and have access to MaxCS features.

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



Logic...	Board Type	Physical ID
0	H323SP	0
1	MobileExtSP	0
2	SIPSP	0
3	HMCP	0@GW01
4	Triton12E	6@GW02
5	Triton12E	5@GW02
6	Triton12E	4@GW02
7	Triton12E	3@GW02
8	Triton12E	2@GW02
9	Triton12E	1@GW02
10	Triton12TGSLs	0@GW02
11	Triton12E	7@GW02
12	Triton12E	8@GW02

Figure 29: In Boards view, open Advanced Configuration panel for SIPSP

2. Add the Firewall/NAT WAN address (in our example, this is **10.40.0.84**) to the Malicious SIP Device List. You do this so that MaxCS will not respond to any SIP messages from that IP address.



SIP Device IP Address
10.40.0.84
204.12.243.82

Figure 30: Add the Firewall/NAT IP address to the Malicious SIP Device list

3. For AltiGen IP phone extension161, make a change that will simulate a loss of connection to MaxCS: On the phone, press **#27**, enter **161** as the extension number, and enter an incorrect password.  
After you do this, the phone will register to the MP-118 gateway and show *LOCAL* on the phone's display.
4. In the AudioCodes configuration tool, set *Automatic Dialing* to 161 instead of 162: Select **VoIP > GW and IP to IP > Analog Gateway > Automatic Dialing**. Refer to [Figure 20](#). 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.
5. Use an AltiGen phone to make an outbound PSTN call starting with trunk access code (for example, 915102520004). The call should be sent out through the FXO ports on the MP11x. 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.
6. Remove 10.40.0.84 from the Malicious SIP Gateway List (see [Figure 30](#)). Add 10.40.0.84 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.
7. Go through the same process to verify that Polycom IP Phone 161 and Analog Phone 167 work correctly. (Note: Polycom and analog phones behind an MP-118 gateway do not need to use #27 to log in during this verification.)

## Limitations

Currently, MaxCS does not support T.38. If you have a FAX ATA box, the box 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 setup instructions.

## AltiGen Technical Support

AltiGen does not provide general configuration support for AudioCodes products. For general configuration information, refer to your AudioCodes documentation. To find your AudioCodes user manual, search for “LTRT-65417 MP-11x and MP-124 SIP User’s Manual Ver 6.6.pdf” in your web browser.

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