MAX Communication Server
Release 8.0

AudioCodes MP-11x Configuration Guide

Intended audience:

AltiGen Authorized Partners

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About This Guide

This guide describes how to configure FXS, FXO, FoIP, PSTN Survivability (SAS), Centrex support, and TLS support for MAXCS 8.0 deployments using an AudioCodes MP-11x or MP-124 gateway.

Please note the following considerations before you begin the procedures in this guide:

- The guide begins with a section General Configuration, which describes configuration steps common to the remaining sections. You must begin with that section; the steps in the remaining sections assume that those configuration changes have already been made.
- Once you have completed the steps in the *General Configuration* section, proceed, in order, to other sections as needed.
- Most examples in this guide illustrate the configuration for an AudioCodes MP-118 gateway. For other models, adjust the port settings as needed.
- AltiGen does not provide general configuration support for AudioCodes devices. Refer to the documentation for your gateway model as needed (to find your user manual, search for "LTRT-65417 MP-11x and MP-124 SIP User's Manual.pdf" in your web browser).

For instructions on configuring the device's Firewall, refer to Article 1242 in the AltiGen Knowledgebase: https://know.altigen.com/questions/1242/Configuring+AudioCodes+Gateway+Firewalls

Requirements

Your system and environment must meet the following requirements:

- You must be using MaxCS Release 8.0.
- You must be using an AudioCodes gateway model MP-11x or MP-124.
- You must have a support agreement with AudioCodes.
- You must have a valid MAXCS third-party SIP license to implement FXS support.
- The gateway must already have a static IP address and must be able to be configured through the web configuration tool.
- The gateway must be running the correct version of firmware: Click the **Home** button and check that you have firmware version 6.60A.265.010.
- MaxCS must have already been properly configured behind NAT (including the port forwarding and the Enterprise Manager settings).
- AltiGen IP phones behind NAT must already be working correctly behind the on-premise firewall.

Note that when you are configuring FXO, MAXCS and the AudioCodes gateway can be on the same LAN or can be inter-connected via VPN. If you are not configuring FXO, you can configure these components over the internet.

Overview

The next figure depicts the configurations modeled in this guide. You will replace our IP addresses with the specific IP addresses for your deployment.

- MAXCS – 10.40.1.43
- Firewall/NAT WAN – 10.40.0.95
- Analog extension – 167
- AudioCodes gateway – 192.168.1.20

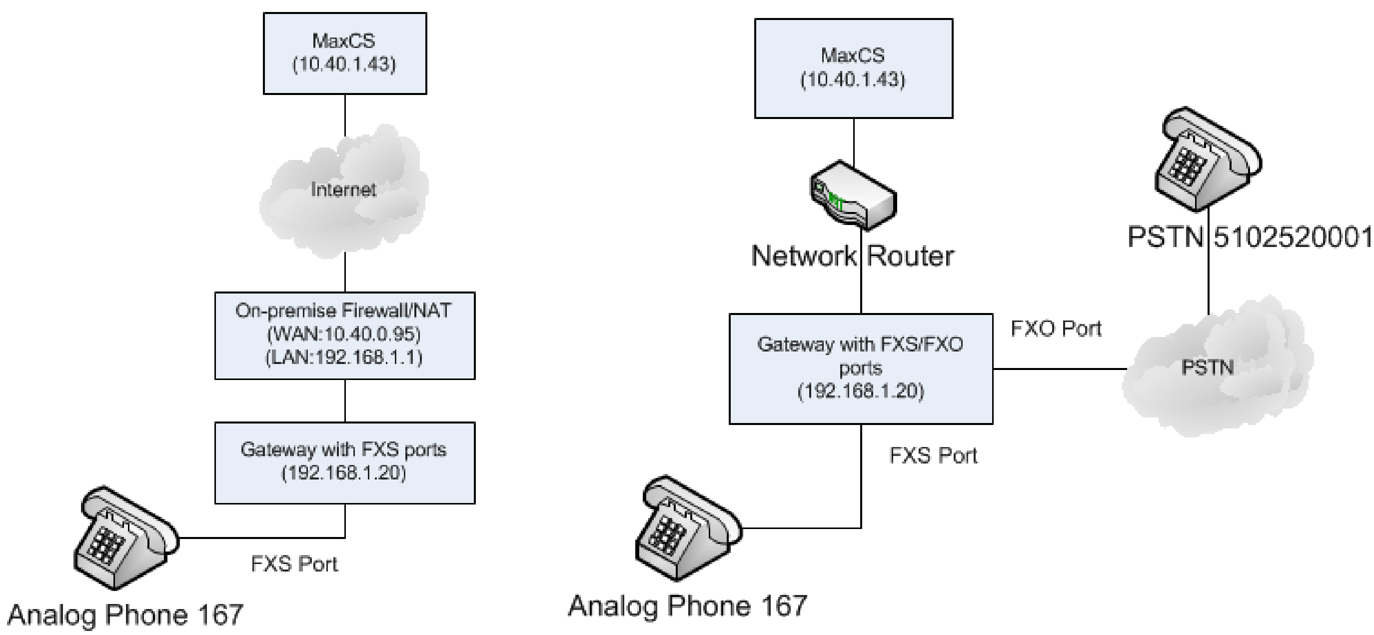


Figure 1: Example configurations for Internet and LAN deployments

In the left diagram, 10.40.1.43 is a public IP address or a NATTED public IP address. In the diagram on the right, 10.40.1.43 is a private IP address.

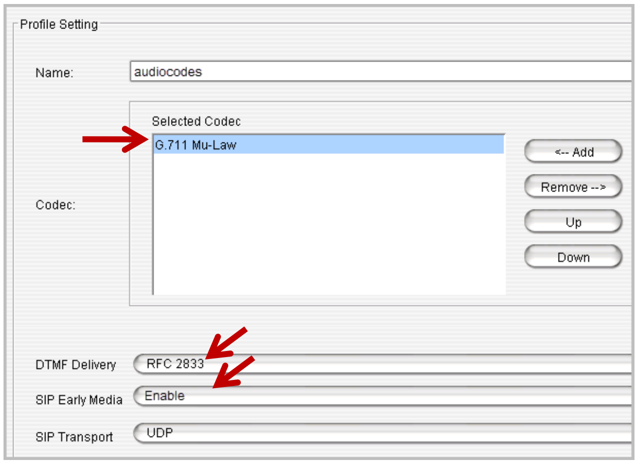
General Configuration

We recommend that you use Hunt Group ID 1 as FXS and Hunt group ID 2 as FXO. This way, in the future, you won't encounter conflicts when you follow this document to add new FXS or FXO configurations.

Complete the steps in this section before you perform any other configuration steps in this guide.

1. Your AudioCodes gateway has many configuration parameters. We recommend that you reset the configuration to the factory default settings before you begin the configuration process. Refer to the instructions in Resetting the Gateway to Default Settings beginning on page 50.
2. Log into MAXCS Administrator and select **VoIP** > **Enterprise Network Management** > **Codec** > **Codec Profile Table**.

3. Add a new profile named *audiocodes*.

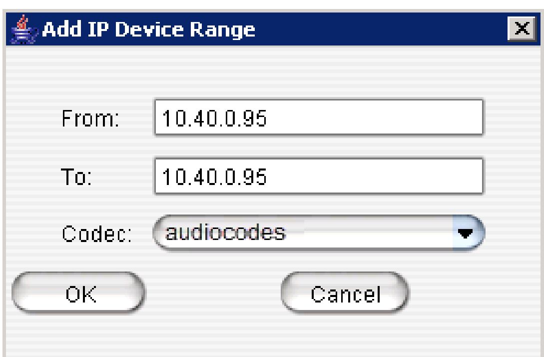


- Assign *Codec* as **G.711 Mu-Law**.
- Set *DTMF Delivery* to **RFC 2833**.
- Set *SIP Early Media* to **Enable**.

Figure 2: Add a new Codec Profile

4. Select **VoIP** > **Enterprise Network Management** > **Servers** > **IP Codec** > **IP Device Range**. Add the Firewall/NAT's public IP address range.

   (If there is no firewall/NAT, then use the gateway's IP address – 192.168.1.20 – in these instructions instead of the firewall's IP address.)



- Set both the *From* and the *To* fields to **10.40.0.95** (the firewall IP address shown in the example on page 4) or the gateway's IP address
- For the *Codec*, select **audiocodes**

Figure 3: Add the Firewall IP address range

5. Open the MaxCS Administrator *Boards* view. Double click **SIPSP**, select **Board Configuration**, and click **Advanced Configuration**. Add **10.40.0.95** to the *Trusted SIP Device List*.

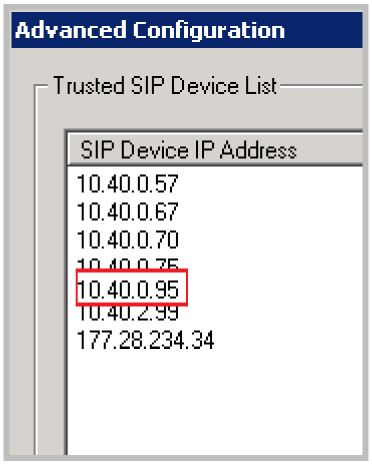


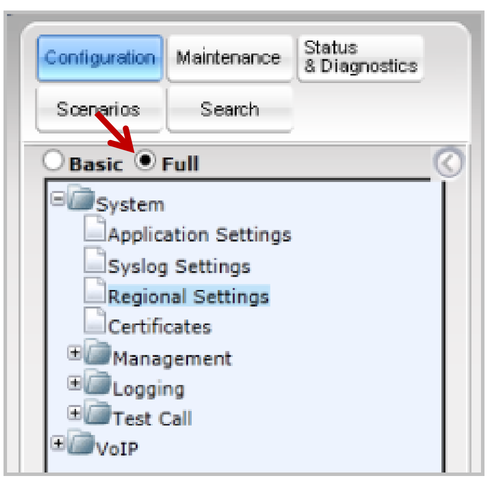
Figure 4: Add the Firewall IP address to the Trusted Device list

If this IP address is not included in the list, then the address will be treated as a malicious SIP device due to excessive SIP messages from that address.

6. Open the AudioCodes configuration tool and log in. (If you see a message about loading scenarios, click **Device Settings** to proceed.)

   (For security purposes, we recommend that you take a minute to change the default AudioCodes configuration tool password now, or do it shortly after you complete the provisioning steps.)

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. Check this setting if you are idle for a few minutes; sometimes the application resets the menus to **Basic** mode.



Click **Submit** after each change, and later click **Burn**, or your changes will be lost after the gateway is restarted.

Figure 5: Set the AudioCodes tool menu to Full

7. Click **Configuration** in the top left corner. Select **System** > **Regional Settings** and make sure that the times are set up correctly.

Some analog phones can display the time on the LCD display. The FXS port will update the analog phone's time while the phone is ringing. If the time is not configured correctly, then the analog phone's time will be incorrect. (The Network Time server can be set up under **System** > **Application Setting**.)

8. Select **VoIP** > **Network** > **IP Routing Table** and confirm that the IP address is 192.168.1.20 (the gateway IP address from the figure on page 4), the prefix length, and the network's gateway IP addresses are set up properly. (This is a read-only page.)

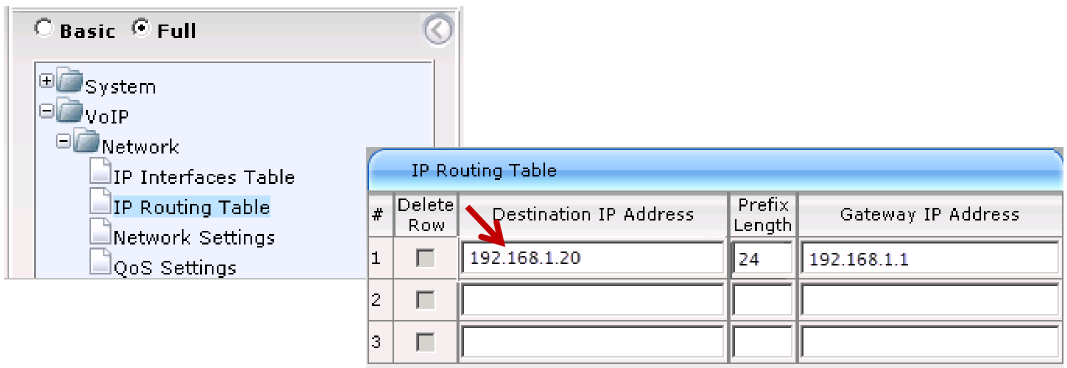
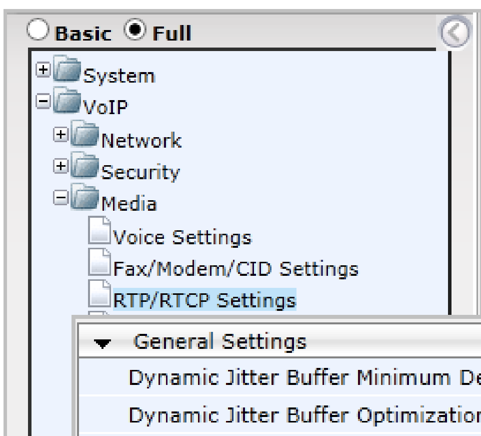


Figure 6: Confirm the network settings

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



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

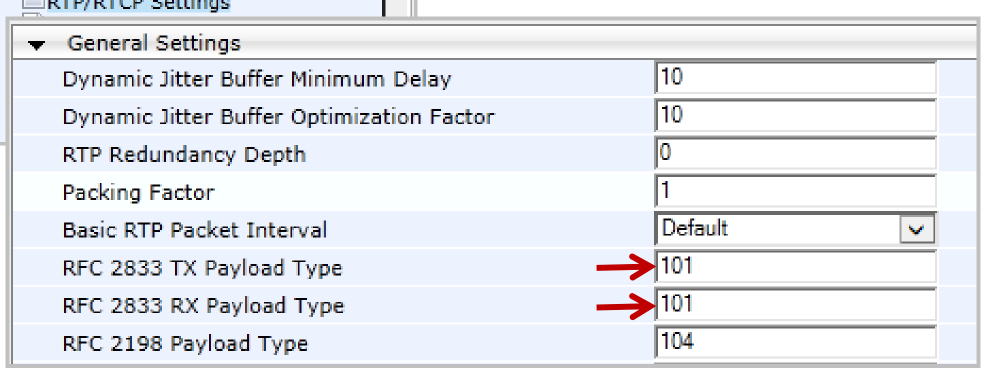
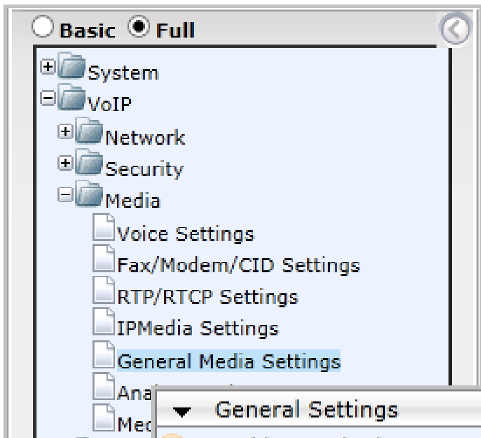


Figure 7: Configure the TX and RX Payload Type settings

10. 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.



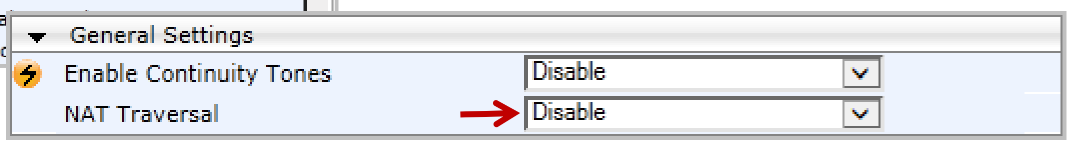
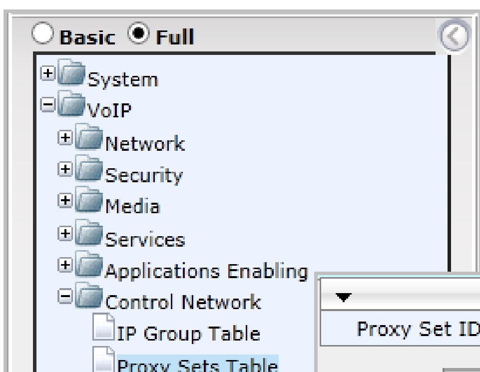


Figure 8: Disable NAT Traversal

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

- Set *Proxy Address 1* to **10.40.1.43:10060** (The MAXCS public IP address and port). Set its *Transport Type* to **UDP**.
- Set *Enable Proxy Keep Alive* to **Disable**. (If later you configure survival settings on FXS ports, you will change this setting.)
- Set *Proxy Keep Alive Time* to **15** (seconds).



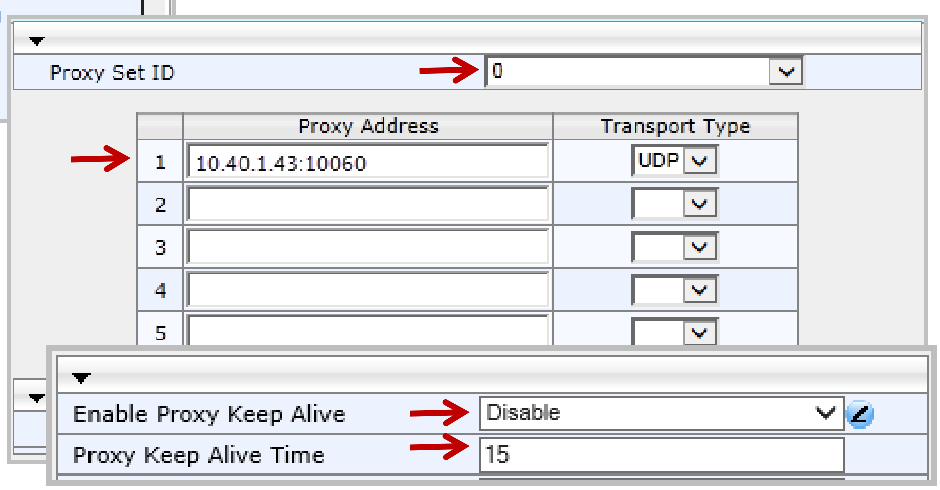


Figure 9: Add an entry to the Proxy Sets table

12. 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**.

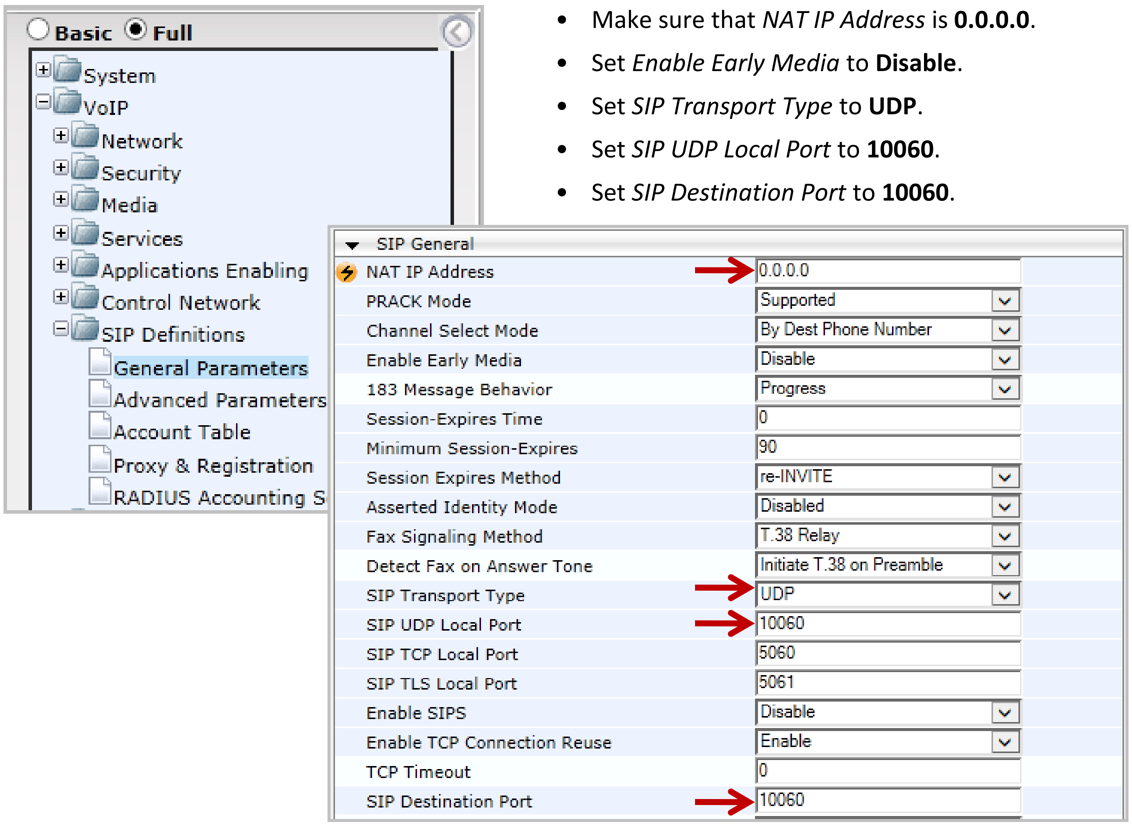
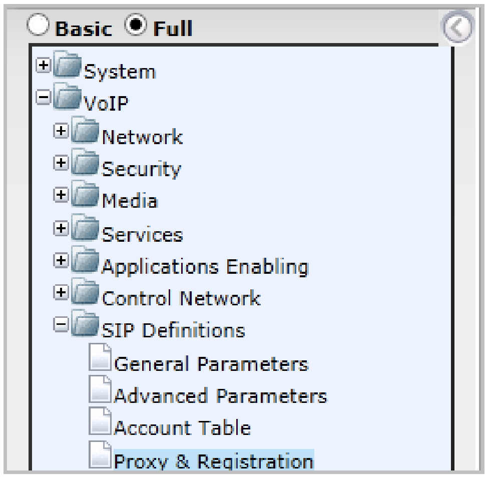


Figure 10: Configure general SIP parameters

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



- Set *Use Default Proxy* to **Yes**.
- Set *Enable Fallback to Routing Table* to **Enable**.
- Set *Prefer Routing Table* to **No**.
- Set *Always Use Proxy* to **Disable**.
- Set *SIP ReRouting Mode* to **Use Routing Table**.
- Set *Enable Registration* to **Enable**.
- Make sure that the *Registrar IP Address* is empty. (It will use the IP address from the Proxy Address/Transport Type under VoIP > Control Network > Proxy Sets Table during registration.)
- Set *Registrar Transport Type* to **UDP**.
- Set *Registration Time* to **100**.
- Set *Re-registration Timing [%]* to **50**.

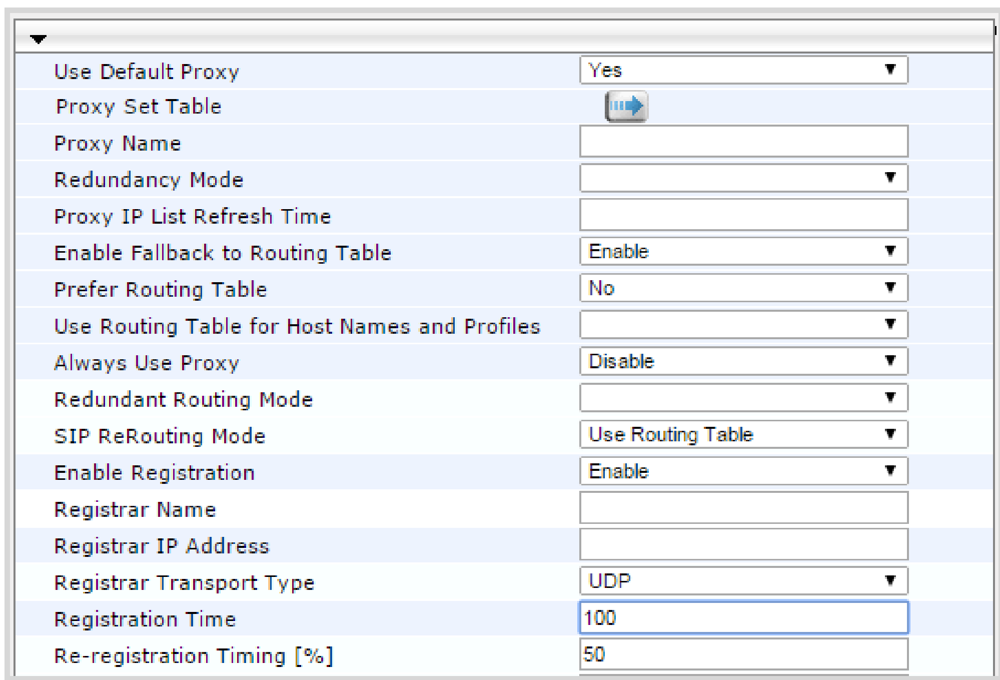


Figure 11: Configure Proxy and Registration parameters

14. Select **VoIP** > **Coders and profiles** > **Coders**. Check that both G.711U-law and G.729 are in the list. (If you choose to configure G.711A-law instead of G.711U-law, make sure that you configure it in the MASCS codec profile.)

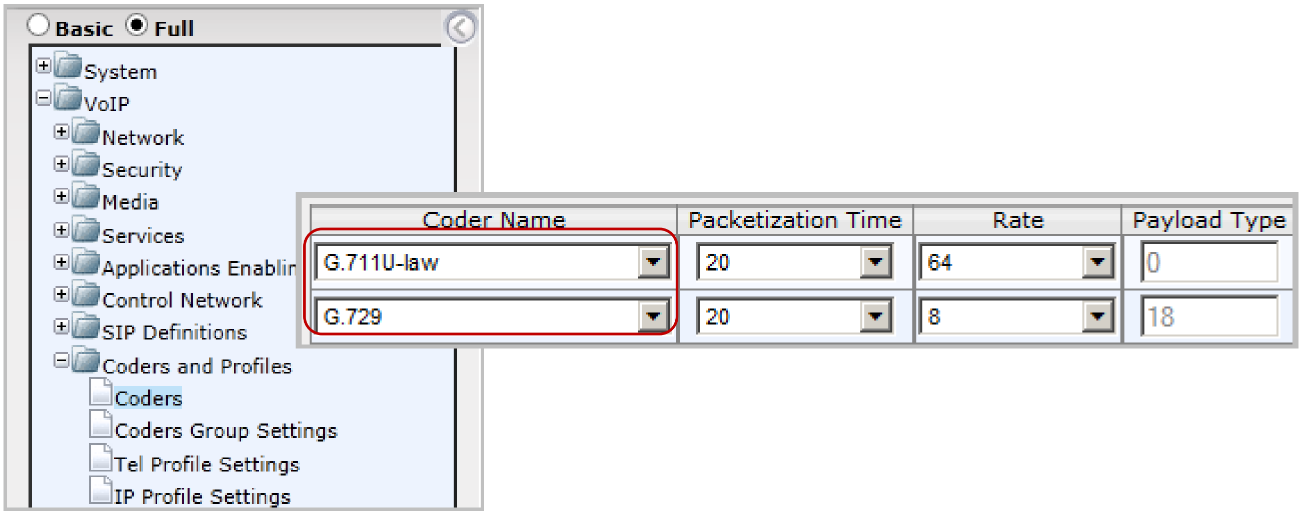
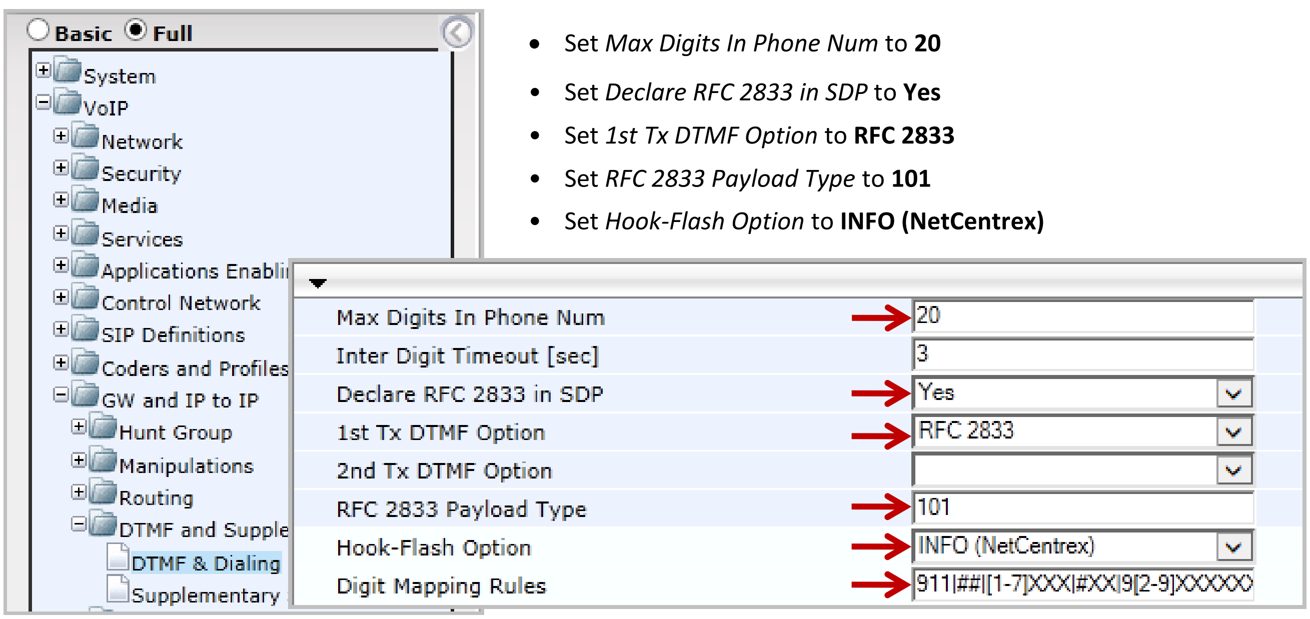


Figure 12: Confirm that G.711U-law and G.729 appear in the Coders list

15. Select **VoIP** > **GW and IP to IP** > **DTMF and Supplementary** > **DTMF & Dialing**. You can omit the *Digit Mapping Rules* field if you are configuring only FXO.

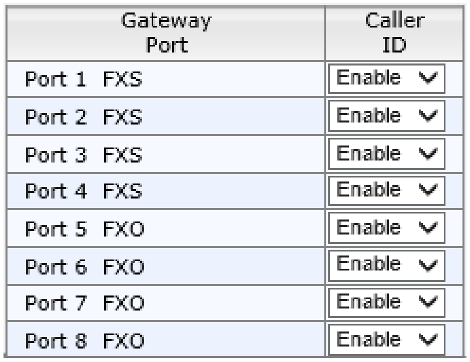
- Set *Max Digits In Phone Num* to **20**
- Set *Declare RFC 2833 in SDP* to **Yes**
- Set *1st Tx DTMF Option* to **RFC 2833**
- Set *RFC 2833 Payload Type* to **101**
- Set *Hook-Flash Option* to **INFO (NetCentrex)**



- If your MAXCS extension length is 3 digits, set *Digit Mapping Rules* to 911|##|[1-7]XX|#XX|9[2-9]XXXXXX|91[2-9]XXXXXXXXX|XX.T
- If your MAXCS extension length is 4 digits, set *Digit Mapping Rules* to 911|##|[1-7]XXX|#XX|9[2-9]XXXXXX|91[2-9]XXXXXXXXX|XX.T

Figure 13: Configure DTMF and Dialing parameters

16. Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **Caller ID Permissions**. Enable Caller ID for the FXS ports and/or for FXO ports.



Gateway Port	Caller ID
Port 1 FXS	Enable
Port 2 FXS	Enable
Port 3 FXS	Enable
Port 4 FXS	Enable
Port 5 FXO	Enable
Port 6 FXO	Enable
Port 7 FXO	Enable
Port 8 FXO	Enable

Figure 14: Enable Caller ID for FXS ports

17. Select **VoIP** > **Media** > **Voice Settings**. Change *DTMF Volume* to **-6**.

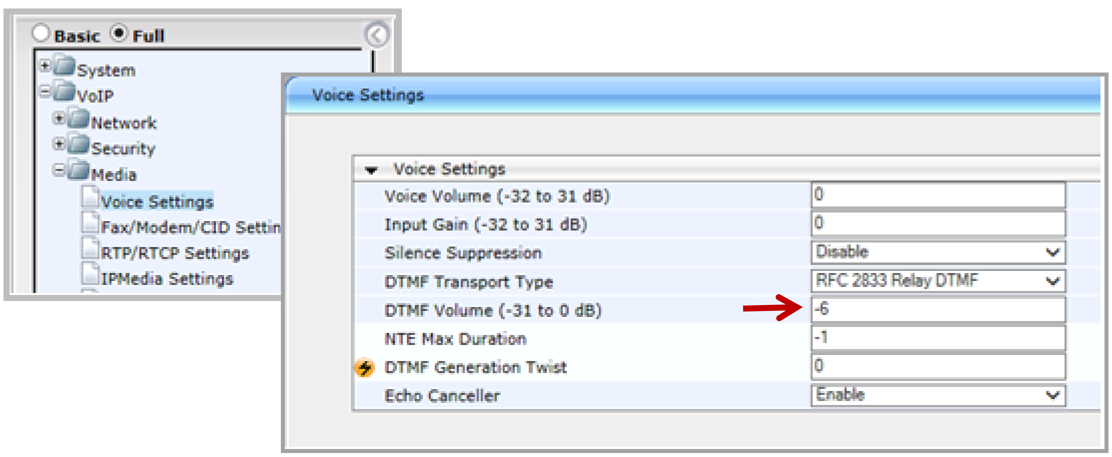


Figure 15: Adjust Voice Settings

18. Submit your last changes. On the toolbar, click **Burn** to save this configuration.

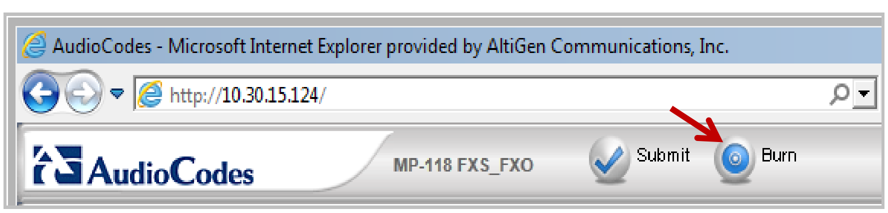


Figure 16: Click Burn to save the configuration changes

19. Click the **Maintenance** button above the menu (see the next figure). Then select **Software Update** > **Load Auxiliary Files**.

At this point, you need to upload two configuration files. These are found in the AltiGen Knowledgebase; they are associated with this article:

- usa_tones_13_NoHold.ini
- usa_tones_13_NoHold.dat

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

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

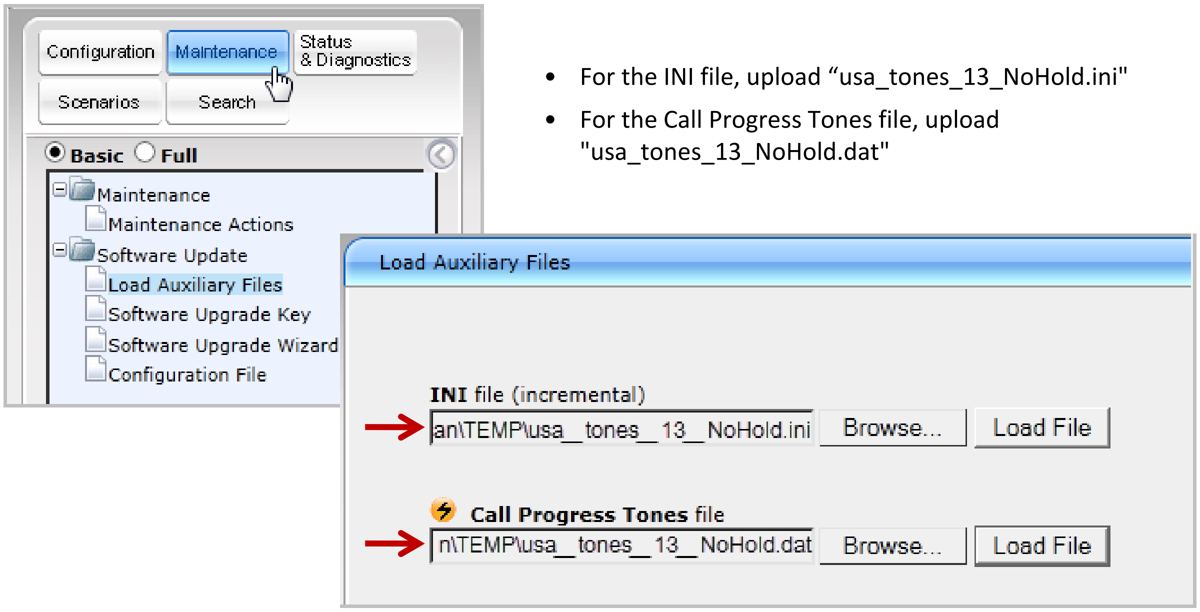


Figure 17: Upload the two data files

20. Submit this last change. On the toolbar, click **Burn**.
21. Restart the AudioCodes gateway.

You have completed the general configuration changes. Continue to a specific procedure, as needed:

- Analog Extension (FXS) Configuration on page 13
- FXO Configuration on page 21
- Fax-over-IP Configuration on page 29
- PSTN Survival Configuration on page 30
- Centrex Configuration on page 46

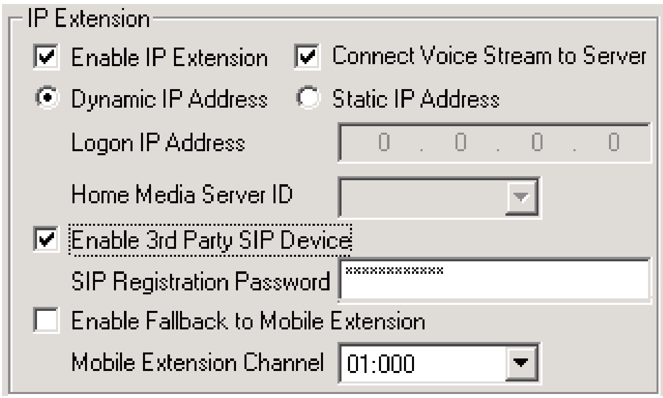
Analog Extension (FXS) Configuration

This section describes how to configure one or more analog extensions using an AudioCodes gateway.

When configured, MAXCS treats analog extensions behind a gateway as third-party IP phones. The analog phones connect to an FXS port. The FXS port converts the analog signal to SIP.

In our example, the range 10.40.x.x is treated as a public an IP address range. The range 192.168.1.x is used as the private IP address range.

1. In MAXCS Administrator, click **PBX** > **Extension Configuration**. Add extension 167 as shown in Figure 1.
2. On the *General* tab, configure the following settings.



- Check *Enable IP Extension*.
- Check *Connect Voice Stream to Server*. (If you clear this checkbox, conferences will fail.)
- Check *Enable 3rd Party SIP Device*.
- Enter a *SIP Registration Password*. We use **5656** in this example.

Figure 18: Configure Extension parameters

3. Select **PBX** > **AltiGen IP Phone Configuration**. For extension 167, clear the checkbox **Enable SIP Telephony Service**.

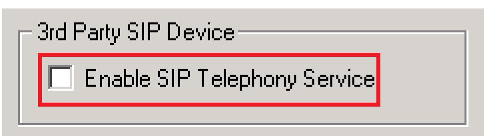
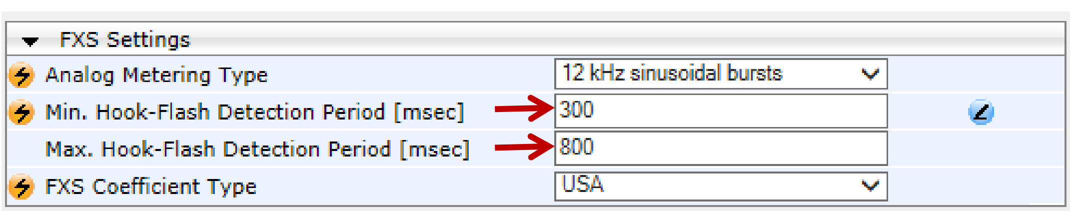


Figure 19: Enable SIP Telephony service

4. In the AudioCodes configuration tool, select **VoIP** > **Media** > **Analog Settings**. Configure the settings as fol-lows.



- Set *Min. Hook-Flash Detection Period [msec]* to **300**.
- Set *Max. Hook-Flash Detection Period [msec]* to **800**.

Figure 20: Configure minimum and maximum Hook-Flash Detection periods

Note: The AudioCodes gateway requires a reboot after you change either of these settings. You can reboot after you complete this configuration, during the last step.

5. Select **VoIP** > **GW and IP to IP** > **Hunt Group** > **Endpoint Phone Number**. Configure entry 1 as follows.

- Set *Channel(s)* to **1**. (This means the first FXS port.)
- Set *Phone Number* to **167** (refer to the extension in Figure 1).
- Set *Hunt Group ID* to **1**.
- Set *Tel Profile ID* to **0**.

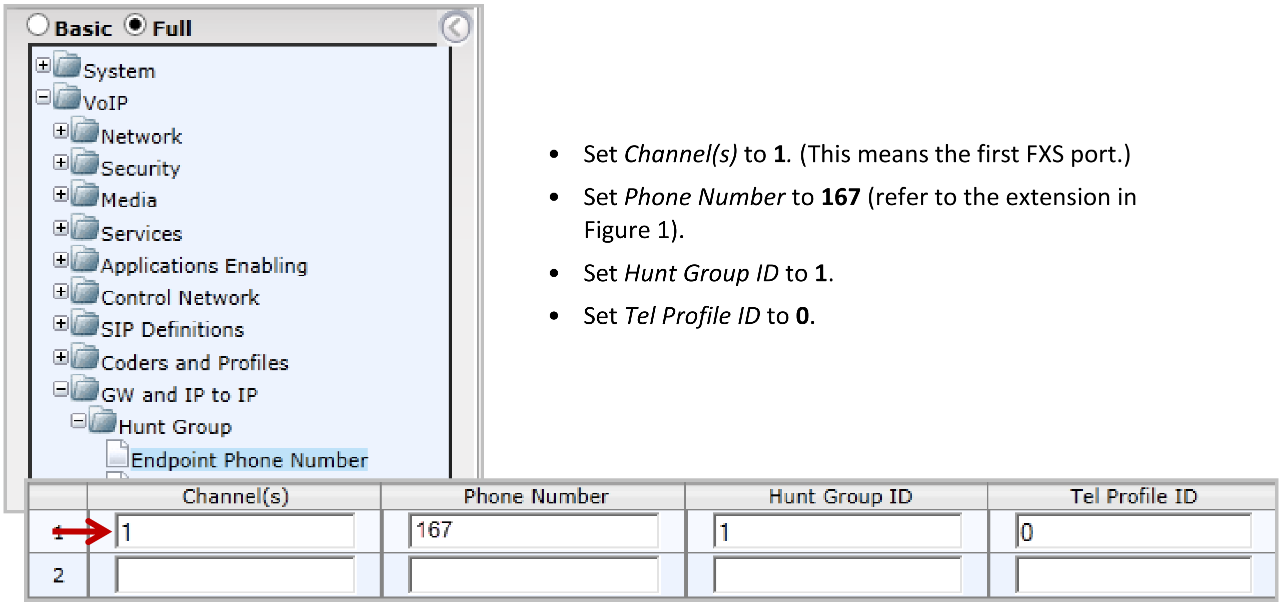


Figure 21: Configure Hunt Group Endpoint number parameters

(Add additional rows if you have additional extensions. *Hunt Group ID* will be **1**, and *Tel Profile ID* will be **0** for additional extensions.)

6. Select **VoIP** > **GW and IP to IP** > **Hunt Group** > **Hunt Group Settings**. Configure entry 1 as follows.

- Enter **1** for the *Hunt Group ID.*
- Set *Channel Select Mode* to **By Dest Phone Number**.
- Set *Registration Mode* to **Per Endpoint**.
- Make sure *Serving IP Group ID* is blank.

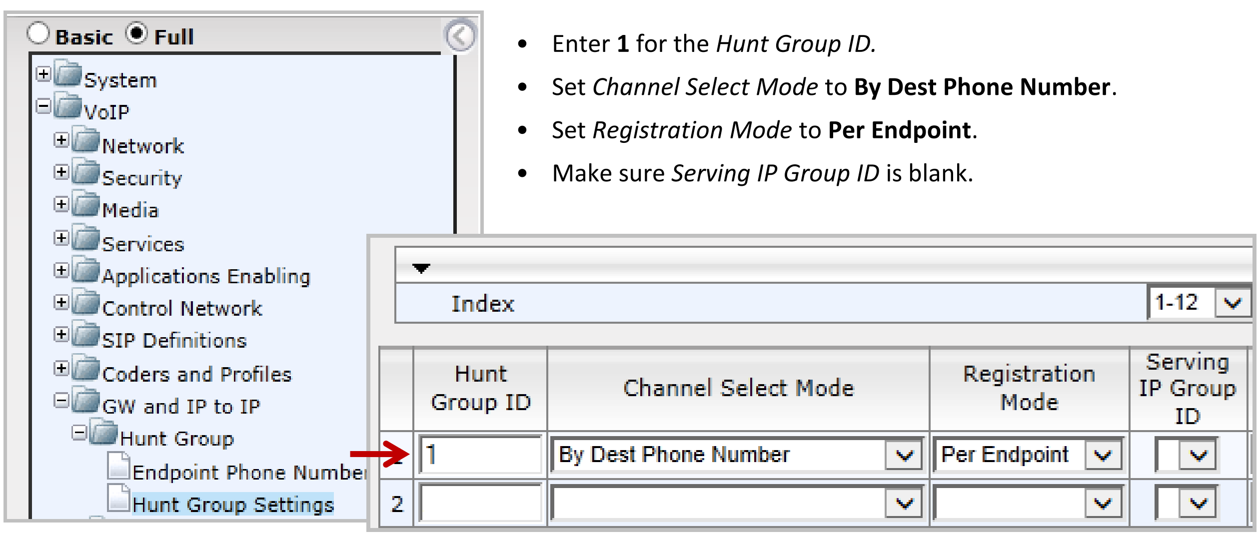
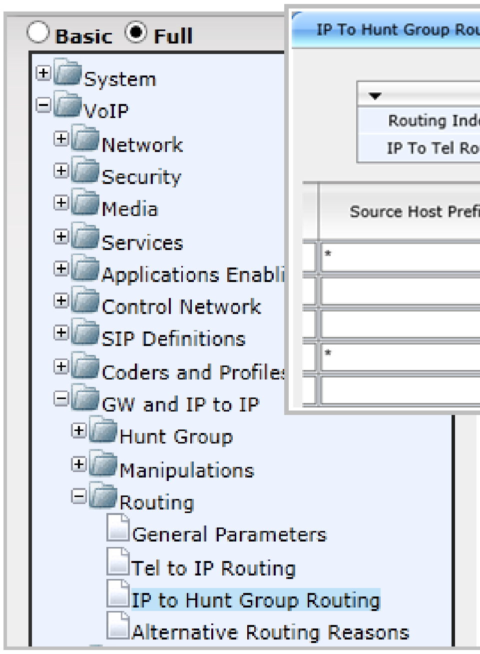
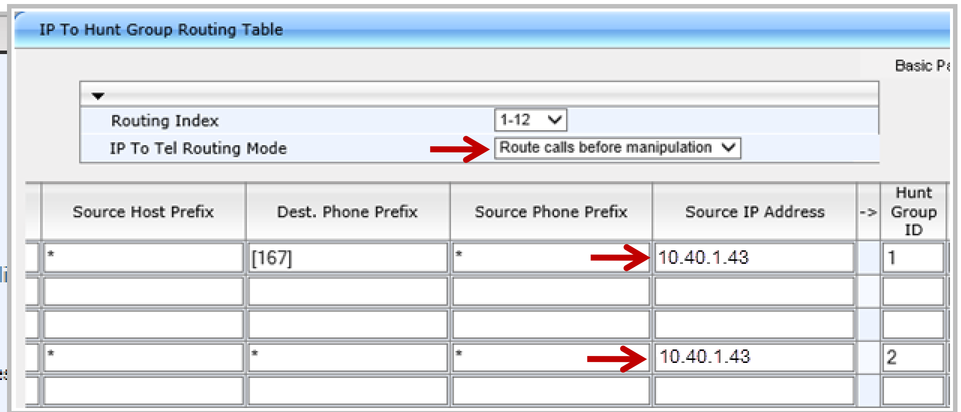


Figure 22: Add Hunt Group ID 1

7. Select **VoIP** > **GW and IP to IP** > **Routing** > **IP to Hunt Group Routing**. Configure entry 1 as follows:





- In the upper panel, set *IP to Tel Routing Mode* to **Route calls before manipulation**.
- Set *Destination Host Prefix, Source Host Prefix,* and *Source Phone Prefix* to an asterisk (*).
- Set *Destination Phone Prefix* to your FXS ports' extension number. In our example, this is **167**. You can enter a range if needed; for example, [167-170] would include extensions 167 through 170.
- Set *Source IP Address* to the MAXCS IP address, **10.40.1.43**.
- Set *Hunt Group ID* to **1**. (This assumes that the FXS port's hunt group ID is 1.)

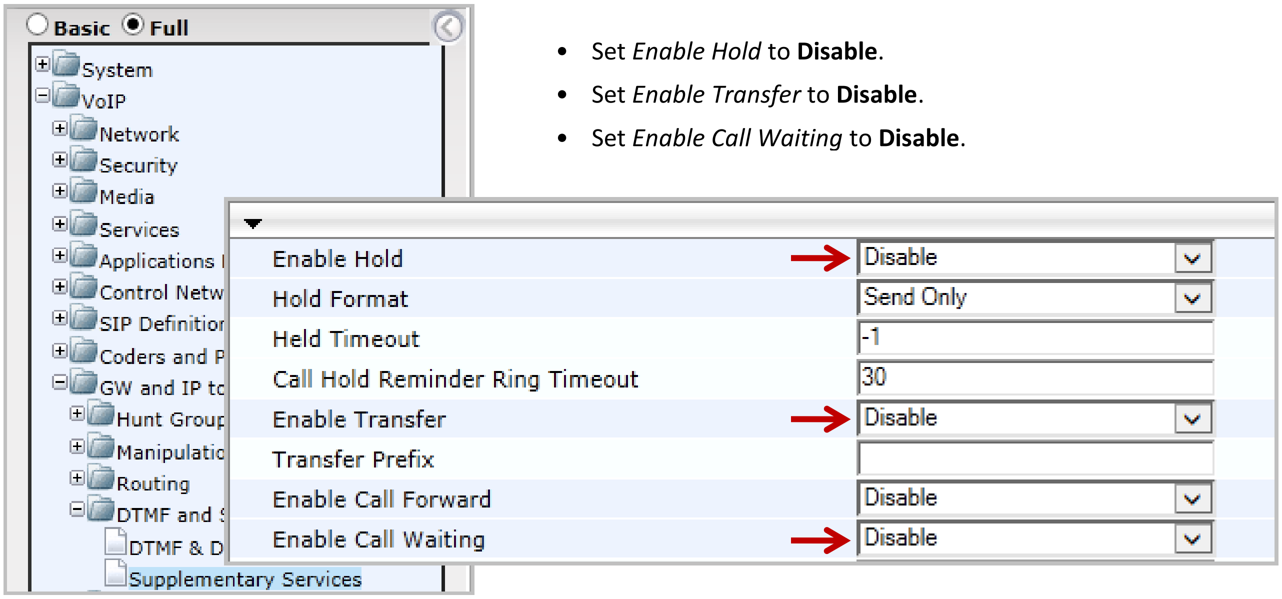
Figure 23: Configure IP to Hunt Group routing rule

Note: When a SIP call is sent to this AudioCodes device, it will match rows from top to bottom. Once it detects is a match, the call will be routed the hunt group of the *first match row* (it will ignore the rest of rows). In the example in Figure 23, if you swap these two rows, then when a call is placed to extension 167, it will be sent to the FXO port instead of the FXS port.

8. Configure another hunt group routing entry; we recommend that you leave a few empty rows after row 1, for future expansion. In our example, we have added the new rule in row 4; refer to the preceding figure.
    - Set *Destination Host Prefix, Source Host Prefix, Destination Phone Prefix,* and *Source Phone Prefix* to an asterisk (*).
    - Set *Source IP Address* to the MAXCS IP address, **10.40.1.43**.
    - Set *Hunt Group ID* to **2** (Hunt Group 2 is used for FXO in this example).

Important: For security reasons, DO NOT use an asterisk into the *Source IP Address* field.

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



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

Figure 24: Configure hold, transfer, and call waiting parameters

10. You can turn on the Message Waiting lamp if your analog phones support MWI. When a user has a new voicemail message, the MWI lamp will flash. Note that enabling the MWI lamp will increase SIP messages for each configured FXS port.

    Skip this step if you do not want to implement this feature.

    On the **VoIP** > **GW and IP to IP** > **DTMF and Supplementary** > **Supplementary Services** page (the same page as the previous step), configure the following settings in the *Message Waiting Indication* section:

- Set *Enable MWI* to **Enable**.
- Set *MWI Analog Lamp* to **Disable**. (Enabling this will introduce 120-150 voltage on the RJ11 cable. If your analog phone uses an MWI analog lamp, then change this option to **Enable**.)
- Set *MWI Display* to **Enable**.
- Set *Subscribe to MWI* to **Yes**.
- Set *MWI Server IP Address* to **10.40.1.43:10060)** (The MAXCS Server IP address, as shown in Figure 1.)
- Set *MWI Transport Type* to **UDP**.
- Set *MWI Subscribe Expiration Time* to **180**.
- Set *MWI Subscribe Retry Time* to **90**.

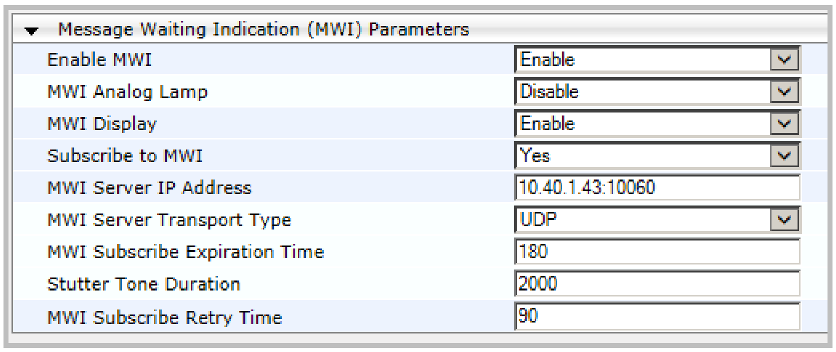


Figure 25: Configure MWI parameters

11. On the same page, set *Enable 3-Way Conference* to **Disable**.

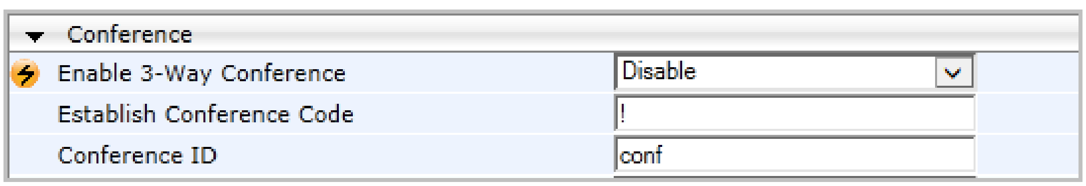
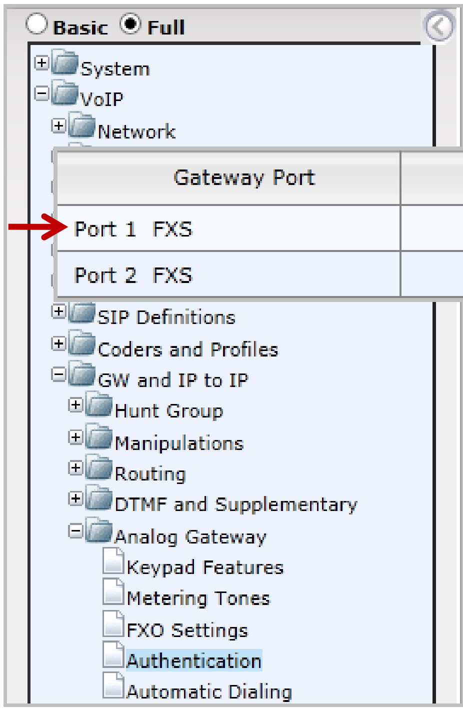
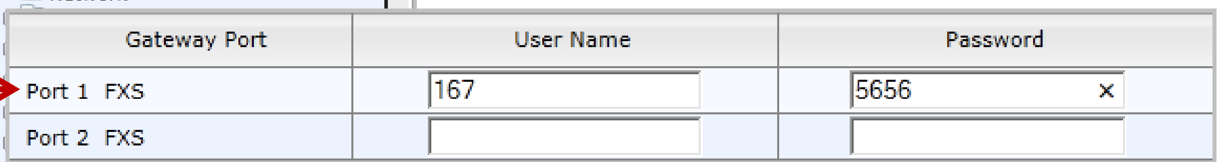


Figure 26: Disable the 3-Way conference parameter

12. Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **Authentication**. Configure *Port 1 FXS* as follows:





- Set *User Name* to **167**.
- For the password, enter **5656**.

(This is the extension and password you set up on page 13.)

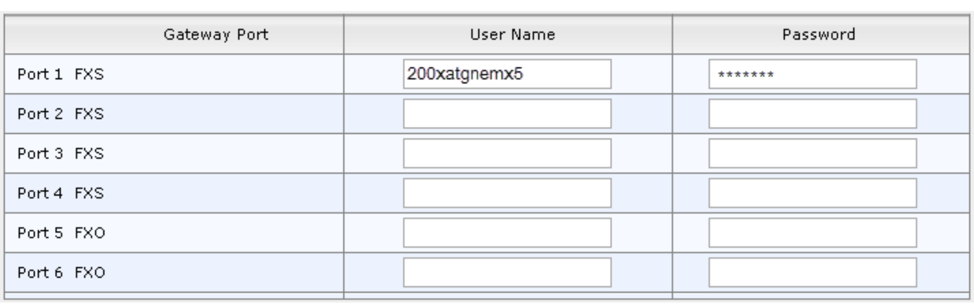
Note: If you save and then restore the .ini file under **Maintenance** > **Software Update** > **Configuration File**, due to AudioCodes' security concern, the password will be erased. You will need to re-enter the password here, after you restore the .ini file.)

Figure 27: Configure Port 1 FXS gateway authentication

13. (Optional) If you want to set up E911 Location ID for relocation for the gateway, select **VoIP** > **GW and IP to IP** > **Analog Gateways** > **Authentication**.

For the appropriate ports, enter the information in the *User Name* field in the following format:

(extension number)xatgnemx(E911 LID)



Gateway Port	User Name	Password
Port 1 FXS	200xatgnemx5	*******
Port 2 FXS		
Port 3 FXS		
Port 4 FXS		
Port 5 FXO		
Port 6 FXO		

Figure 28: Configure e911 Location ID

For example, to specify extension number 200 assigned to Location ID 5, enter the following string:

```
200xatgnemx5
```

Enter the extension's SIP Registration password (from MaxCS Administrator) in the *Password* field (refer to the chapter *Setting up IP Extensions* in the *MAXCS 7.5 Administration Manual*.

For details on configuring E911 Location IDs for IP phones, refer to the chapter *Location-Based E911 (for Re-location)* in the *MAXCS 7.5 Administration Manual*.

14. To verify that the settings are correct and that the phones work correctly, select **VoIP** > **GW and IP to IP** > **Hunt Group** > **Endpoint Phone Number**. This table defines phone numbers for gateway endpoints.

15. Click the **Register** button.

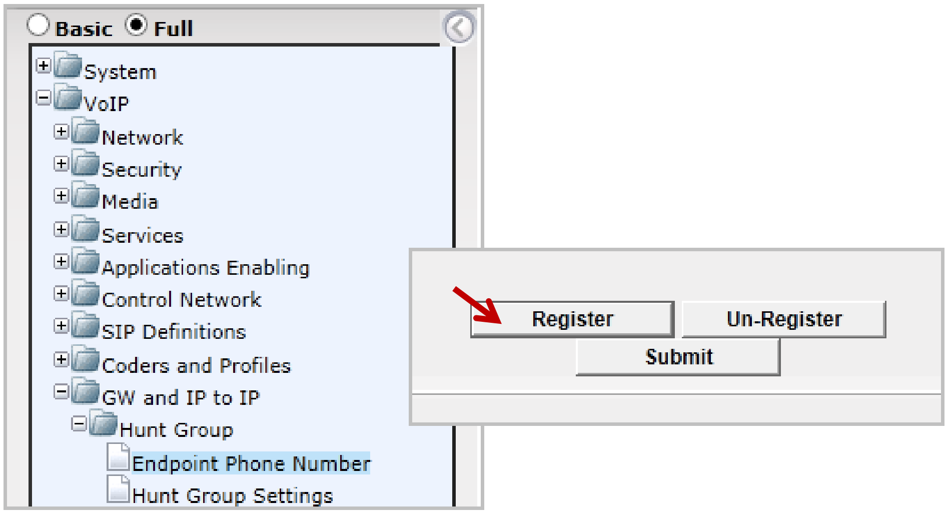


Figure 29: Click Register

16. Log into MAXCS Administrator. Open the Extension view; extension 167 should be listed there.

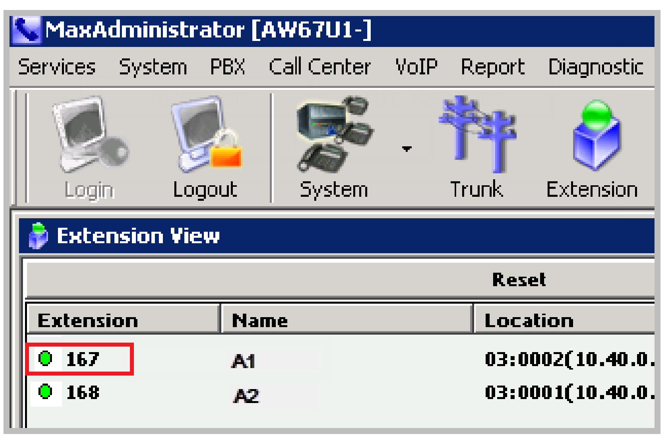


Figure 30: Confirm that extension 167 appears in Extension View in MAXCS

17. Attach an analog phone to the first FXS port and make a few calls to test that everything works correctly.

18. Submit this last change. On the toolbar, click **Burn**.

19. Restart the AudioCodes gateway.

FXO Configuration

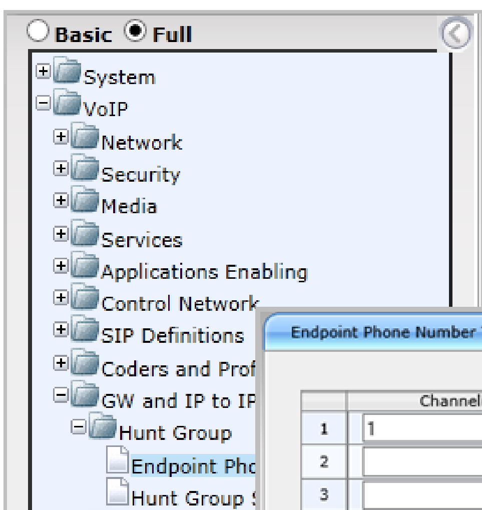
FXO port configuration is supported on **premise-based** MAXCS systems. It is not supported on MAXCS Private Cloud deployments.

Before you begin these procedures, make sure that you have completed all of the steps in the first section, General Configuration, beginning on page 4. If you are configuring only FXO, then you may omit the entries in the Digit Mapping field as mentioned on page 11.

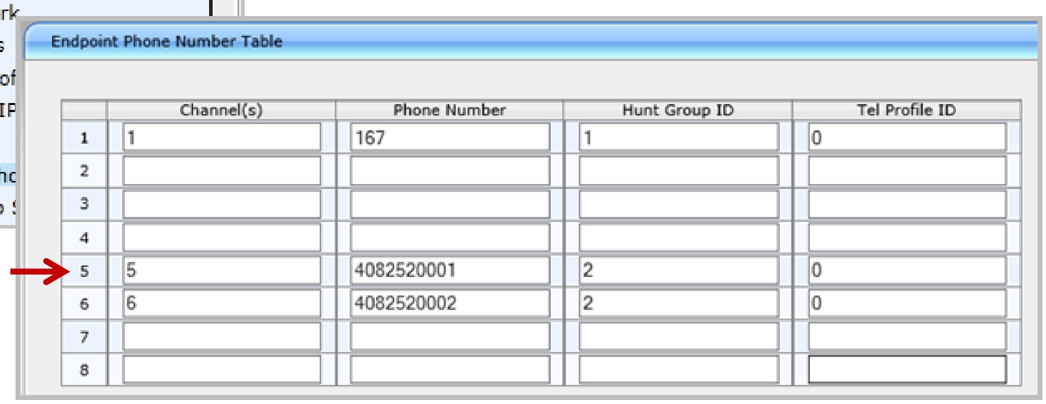
Your environment must meet the following requirements:

- Your system must be running MAXCS Release 7.5.0.502 or later on an enterprise (on-premise) deployment. These configurations **are not supported on a MAXCS Private Cloud deployment**.
- No NAT is allowed between MAXCS and the gateway. For example, in Figure 1, IP addresses 10.40.1.43 (MAXCS) and the gateway (192.168.1.20) must be routable bi-directionally.
- MAXCS and the gateway can be on the same network.

1. In the AudioCodes configuration tool, click **Configuration** in the top left corner. Make sure that **Full** is selected for the menus.
2. Select **VoIP** > **GW and IP to IP** > **Hunt Group** > **End Point Phone Number**. In addition to any extension numbers that may have already been configured, add trunk numbers. Note that empty rows between the FXS and FXO channels do not matter.



- Set entry 5 to channel **5**. (On the MP-118, this is the first FXO port.)
- For channel 5, set the *Phone Number* field to your PSTN phone number (in our example, this is 4082520001) and set *Hunt Group ID* to **2** (FXO group). *
- Add more channels as needed; in this example, we have also configured channel 6.


Endpoint Phone Number Table

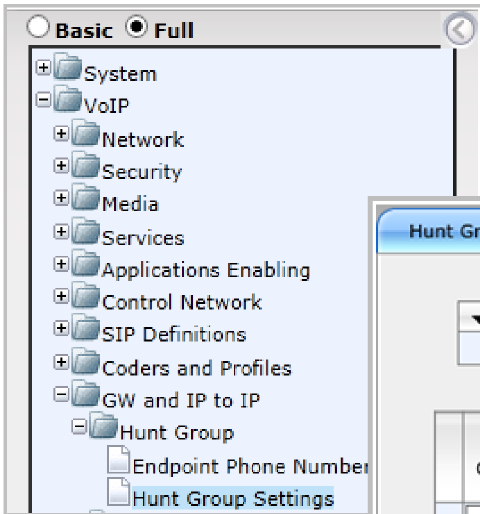
	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	167	1	0
2				
3				
4				
5	5	4082520001	2	0
6	6	4082520002	2	0
7				
8				

Figure 31: Add trunk numbers to the Endpoint parameters

*For an incoming call to the FXO port, if a caller ID is not received, this number, 4082520001, will be used as the default caller ID. The gateway will forward the call to MaxCS with the caller ID through SIP.

3. Select **VoIP** > **GW and IP to IP** > **Hunt Group** > **Hunt Group Settings**. Configure a hunt group ID 2 as follows.

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



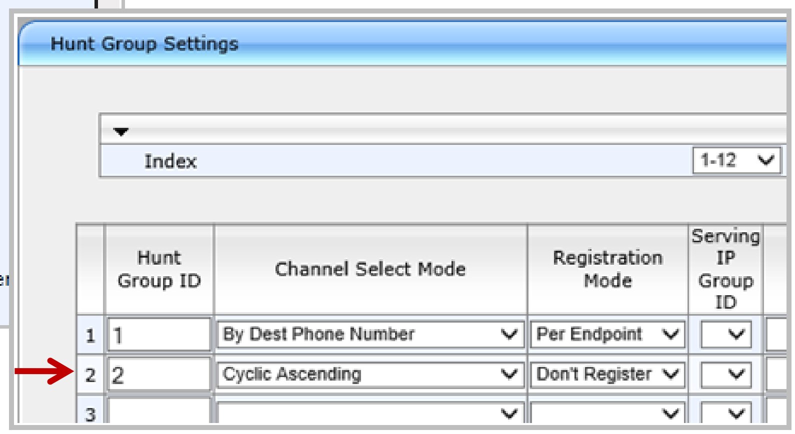
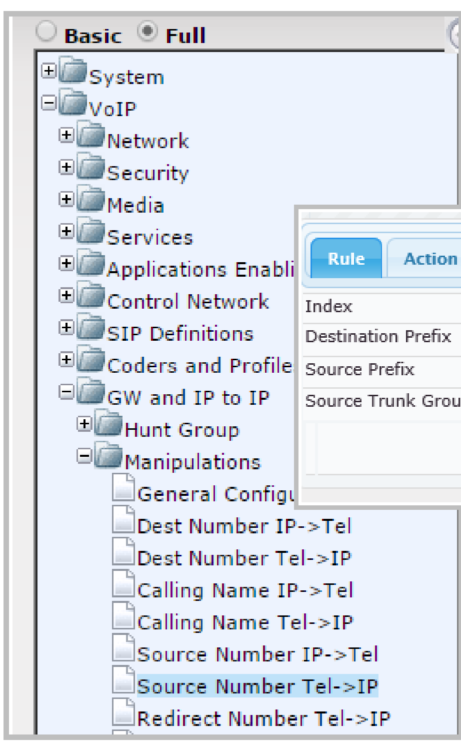


Figure 32: Configure Hunt Group 2

4. Select **VoIP** > **GW and IP to IP** > **Manipulations** > **Source Number Tel -> IP**. Click **Add** to add a new rule.

- On the **Rule** tab, set *Source Trunk Group* to **2**.
- On the **Action** tab, set *Prefix to Add* to **TRK** (use uppercase letters). This adds a "TRK" prefix to the caller ID, so that MAXCS or AudioCodes will recognize that this call is a trunk call. This is necessary because the IP address is shared by FXS and FXO ports.



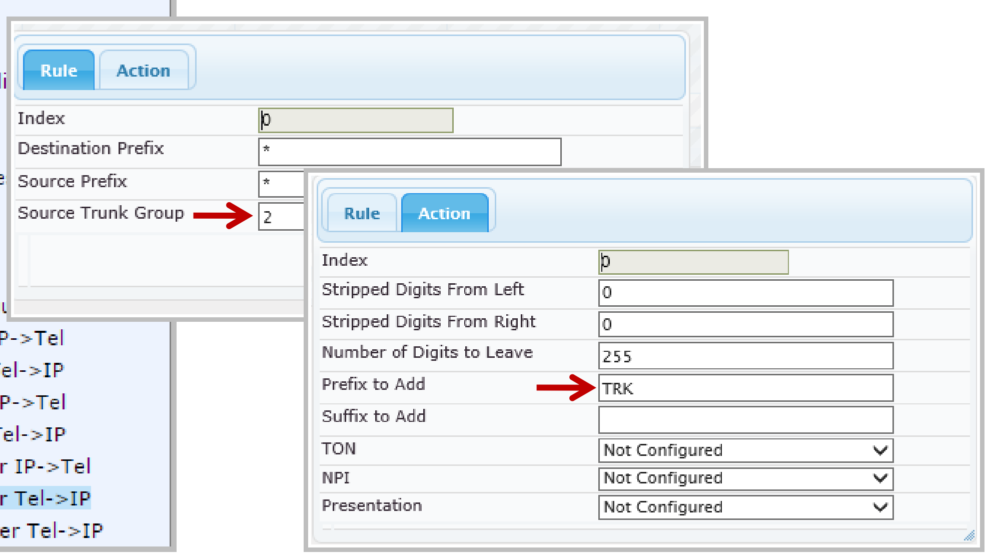


Figure 33: Configure source number to IP number manipulations

5. Select **VoIP** > **GW and IP to IP** > **Routing** > **IP to Hunt Group Routing**. Configure a new entry as follows:

- Set *Dest. Host Prefix, Source Host Prefix, Dest. Phone Prefix,* and *Source Phone Prefix* all to asterisk (*).
- Set *Source IP Address* to the MAXCS IP address, **10.40.1.43**.
- Set *Hunt Group ID* to **2**.

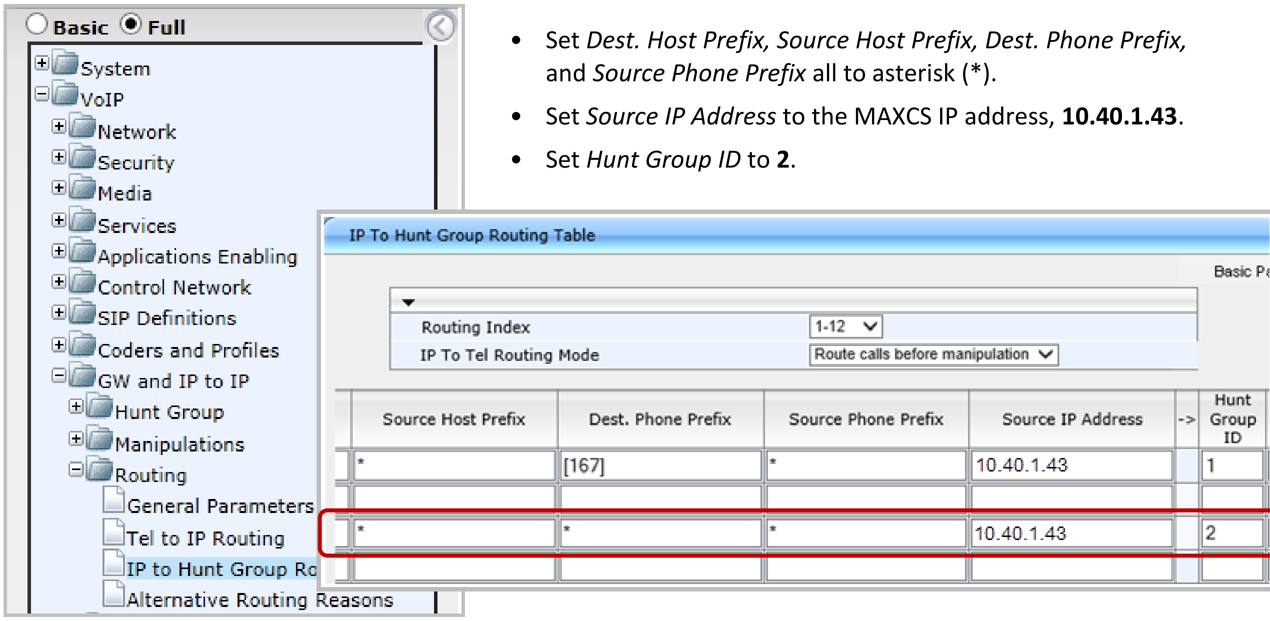


Figure 34: Configure a new IP to Hunt Group routing rule

This rule specifies that if the number sent by MAXCS is not the extension number, then treat the number as a PSTN number and route it to Hunt Group 2 (FXO group).

6. Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **FXO Setting**. Set *Dialing Mode* to **One Stage**; set *Discon-nect on Dial Tone* to **Enable**.

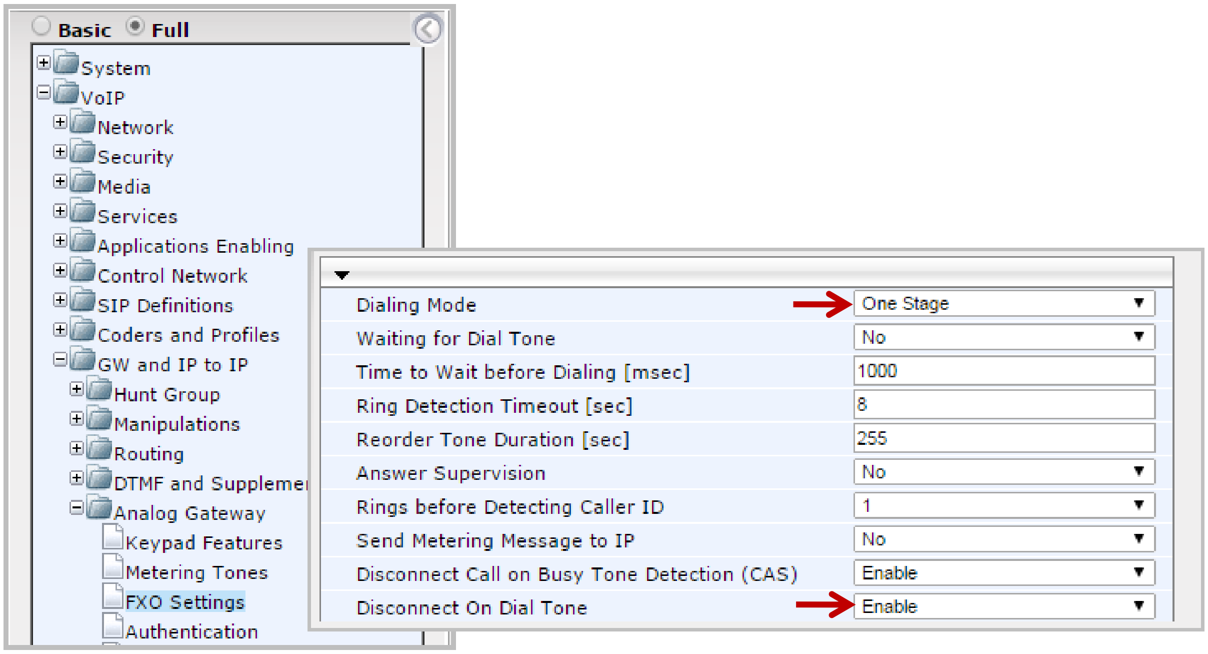


Figure 35: Configure FXO parameters

7. Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **Automatic Dialing**. Set *Port 5* (the first FXO port) to **167**.

   This specifies that when a PSTN call comes into the gateway's FXO port, the gateway will send an IP call with the destination DID or DID suffix 167 (when the SIP Trunk receives a number, it will be interpreted as a DID number or a DID suffix instead of an extension number).

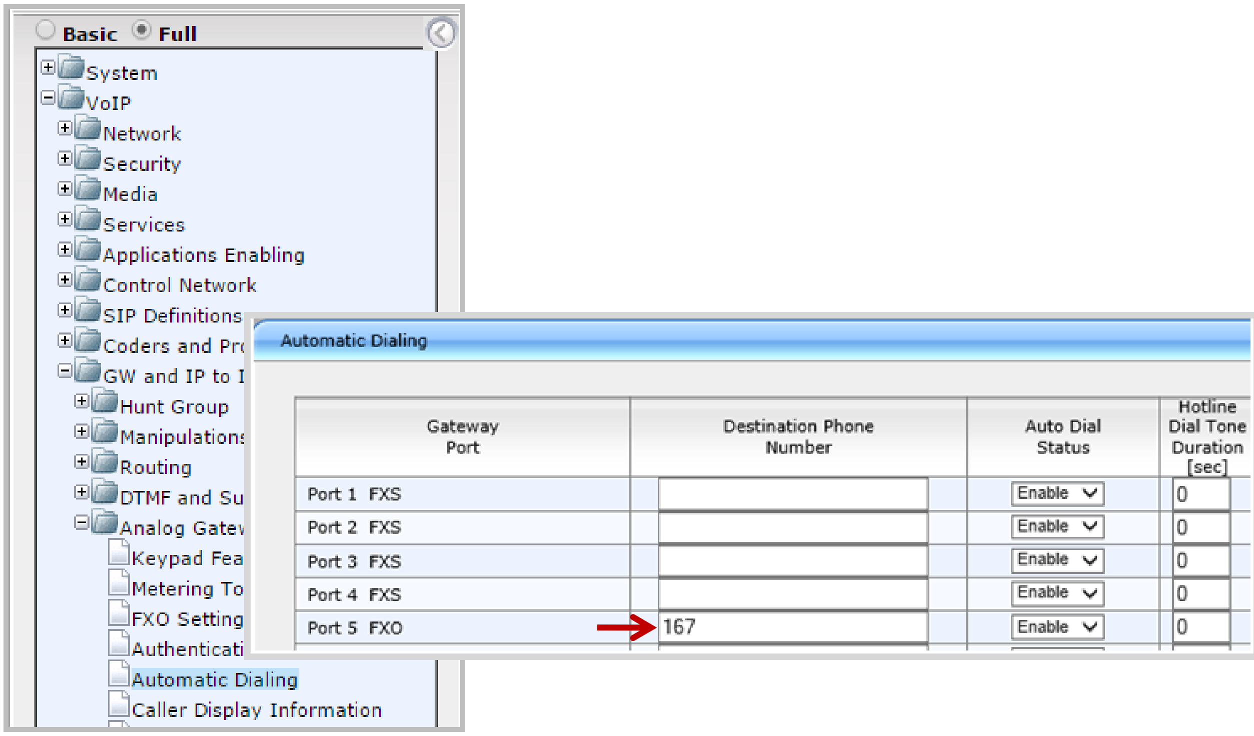


Figure 36: Configure automatic dialing rules for port 5

If you have more FXO ports, then you need to enter more numbers in the *Destination Phone Number* fields. The number can be different, or can be 167.

8. Submit your last changes. On the toolbar, click **Burn** to save this configuration.

9. Next, configure a SIP Group with SIP servers. Log into MAXCS Administrator and double-click **SIPSP** in *Boards* view. Click **Board Configuration** and then click **SIP Group Configuration**.

Note: If you are setting up multiple MP-1xx gateways for FXO lines, then you must create one Group for each gateway. The trunks may all be included in the same Out Call Route or have the same Trunk Access Code. You must also assign the number of SIP trunks to that group to match the number of FXO lines on the gateway.

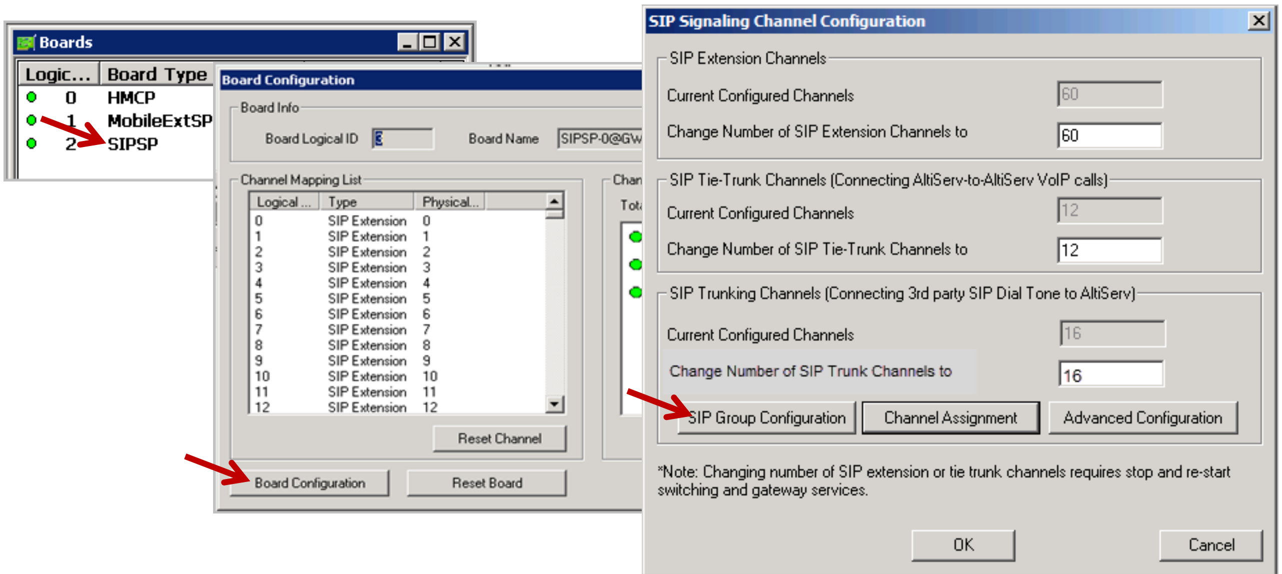


Figure 37: Open the SIP Group Configuration panel

10. Click **Add**. Enter a name for the group and click **OK**.

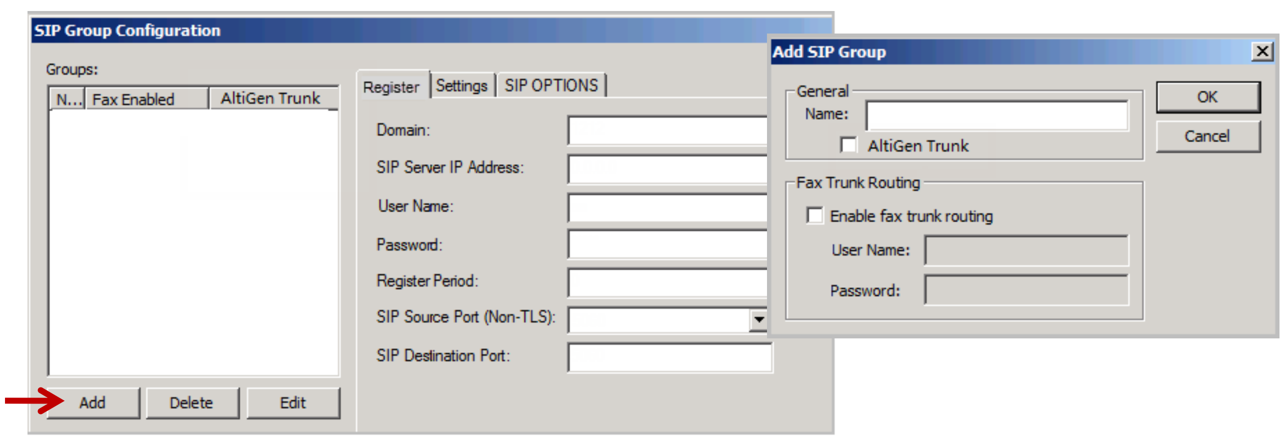


Figure 38: Enter a name for this new SIP Group

11. You will now add a SIP server to this new SIP Group.

In the panel, highlight the new SIP Group (in the next figure, the new group is named *mysip*) and click the lower **Add** button (the one that is below the SIP Servers list). Enter the IP address of the AudioCodes gateway for the domain and click **OK**.

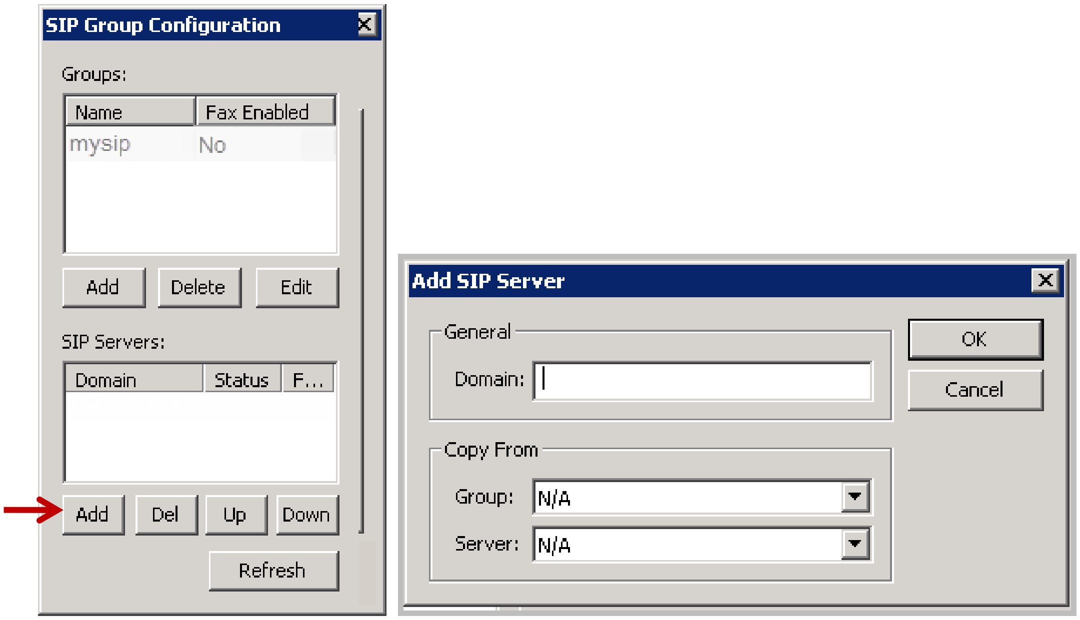


Figure 39: Add a SIP server to the new SIP Group

12. Highlight the new SIP Group. Select this new SIP Server and configure the settings on the *Register* tab:
    - For the SIP Server IP Address, enter the IP address of the AudioCodes gateway
    - For *User Name,* enter **audiocodes**
    - Leave the *Password* field blank
    - For *Register Period*, enter **0**
    - For the SIP Source Port (Non-TLS), enter **10060**
    - For the SIP Destination Port, enter **10060**

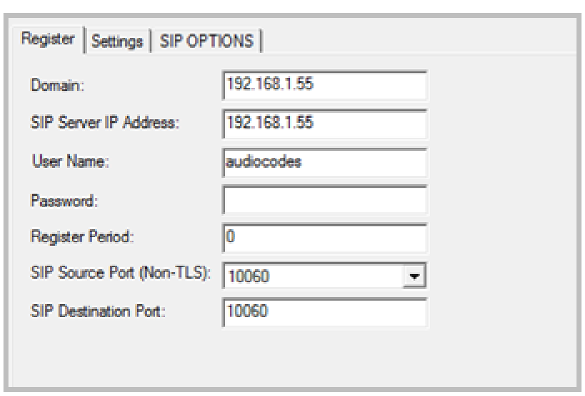


Figure 40: Configure SIP Trunk parameters

13. Switch to the *Settings* tab. Change *SIP Protocol Field* to **FROM Header**. Click **Apply**.

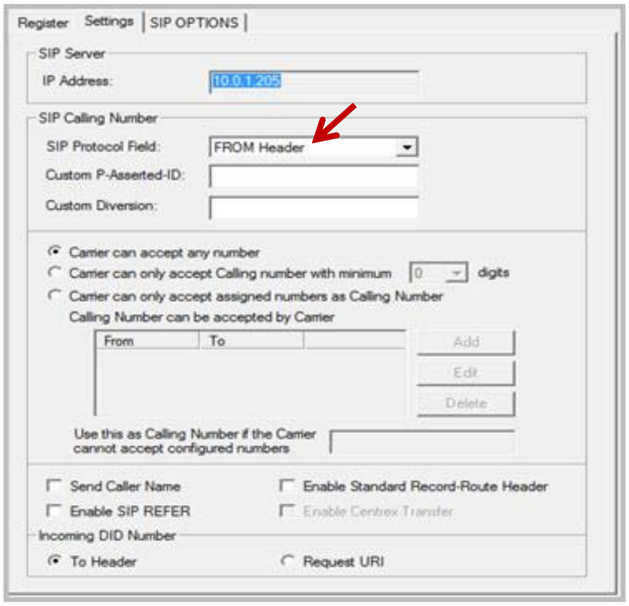


Figure 41: Change the SIP Protocol Field

14. Next, enable channels. Open the *Board Configuration* panel. Click **Channel Assignment**. Select the appropriate channels (use **Ctrl-Click** to select multiple channels) and click **Assign Group**. Choose the SIP Group you created earlier and click **OK**.

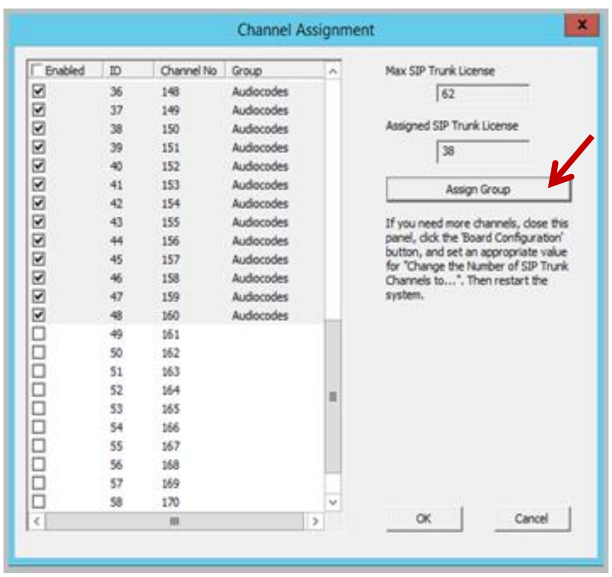


Figure 42: Enable channels

15. Now you can add these newly-created trunks to an outbound route or assign them to a trunk access code.

16. At this point, test that you can make outbound calls and inbound calls from MAXCS using your IP phone and an FXS port. Do not continue to the next section until these calls are working correctly.

Fax-over-IP Configuration (Optional)

This section describes how to configure Fax-over-IP (FoIP) with the AudioCodes gateway where the Fax device is attached to the AudioCodes gateway's FXS port.

SIP-Trunk side FoIP is supported only on AltiGen SIP Trunks. SIP-Trunk side FoIP is not supported on AudioCodes FXO or PRI/T1 channels.

Sending and receiving faxes over IP service has known limitations. AltiGen, along with many other companies, uses the T.38 industry standard for FoIP configuration. The T.38 protocol contains minor variations in how it can be implemented. Because of these variations, one provider's FoIP handling can vary from another's, thus introducing the possibility of incompatibilities. As the standard continues to evolve, it is reasonable to expect these variations to diminish over time.

If your organization typically sends frequent faxes that are lengthy multiple page documents, consider retaining a few analog lines and traditional fax machines as a backup option.

Before you begin, make sure that you have completed the procedures in the section General Configuration beginning on page 4.

To configure Fax-over-IP (FoIP) support, follow these procedures.

1. Perform all of the steps in the section Analog Extension (FXS) Configuration.
2. Follow the steps in the FoIP chapter of the *MAXCS 7.5 ACM Administration Manual* to configure MAXCS.
3. Open the AudioCodes web configuration tool and log in.

Important! Remember to set the menu to **Full** so that you can see all menu commands.

4. On the left panel, select **VoIP** > **SIP Definitions** > **General Parameters**.
   - Set *Fax Signaling Method* to **T.38 Relay**
   - Set *SIP UDP Local Port* to **10060**

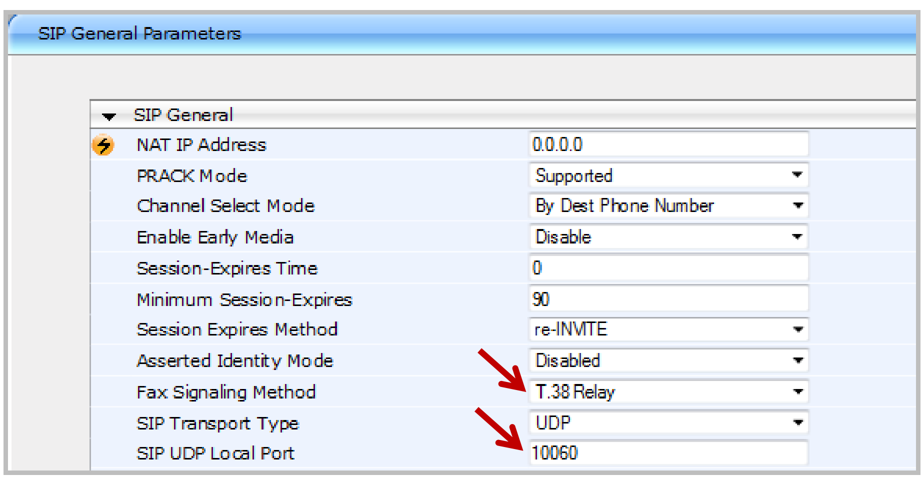


Figure 43: Configure SIP general parameters

5. On the left panel, select **VoIP** > **Media** > **Fax/Modem/CID Settings**.
    - Set *Fax Transport Mode* to **T.38 Relay**
    - Set *Fax CNG Mode* to **Sends on CNG or v8-cn**
    - Set *CNG Detector Mode* to **Relay**
    - Set *Fax/Modem Bypass Coder Type* to **G711 Mulaw**

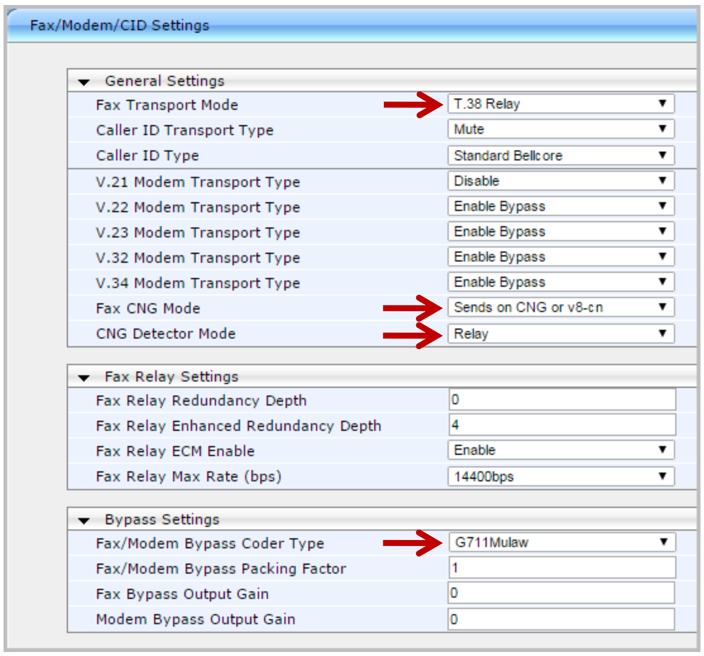


Figure 44: Set Fax parameters

6. Submit these changes. On the toolbar, click **Burn** to save this configuration.
7. Reboot your AudioCodes device so that all of the changes take effect.
8. After the device has rebooted, send and receive a fax to that extension to confirm that the configuration is correct and that faxes can be sent and received.

PSTN Survival (SAS) Configuration (Optional)

This section describes how to configure the gateway so that if the gateway loses connection to MAXCS, calls from PSTN will loop back to the AudioCodes device's SAS service (Stand Alone Survivability service) and route calls to an FXS port or to an IP extension.

Calls from the FXS port or IP extension are managed by the SAS service. An SAS service is a simple softswitch service that performs basic inbound, outbound, and extension-to-extension calls. It can serve as an emergency softswitch while the MaxCS server is not reachable. The SAS service will route calls to the FXO port or to the designated IP extension, based upon the number the caller dialed.

From the SAS perspective, FXS ports on AudioCodes devices are considered IP extensions.

Note: An AudioCodes MP-1xx SAS service can handle up to 25 concurrent IP phone registrations.

PSTN SAS General Configuration

You will begin this configuration with this first section. Once you have completed these steps, you will then continue on to either the LAN configuration section or the Cloud configuration section.

Before you begin, make sure that you have completed the steps in the preceding sections of this guide. Make sure that the menus in the configuration tool are set to **Full**.

1. Select **VoIP** > **Applications Enabling** > **Applications Enabling**.
2. Set *SAS Application* to **Enable**.
3. On the toolbar, click **Submit**.

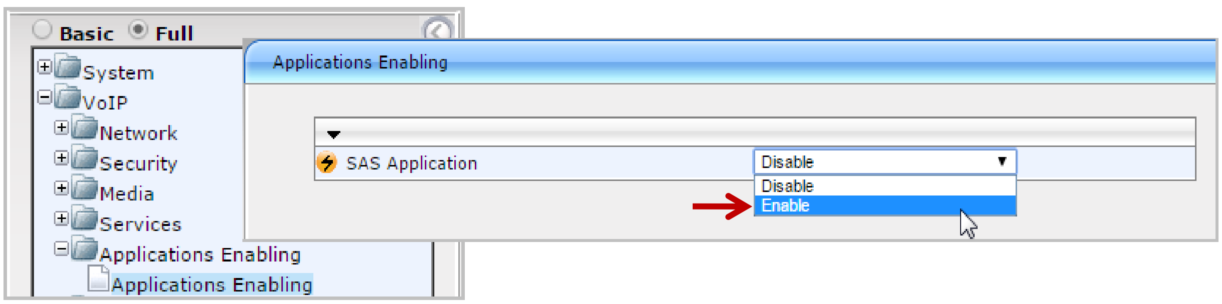


Figure 45: Enable the SAS application

4. On the toolbar, click **Burn** to save this configuration.
5. Restart the MP-11x device.

Follow these steps after the device has restarted:

1. Log back into the configuration tool and select **VoIP** > **Control Network** > **Proxy Sets Table**.

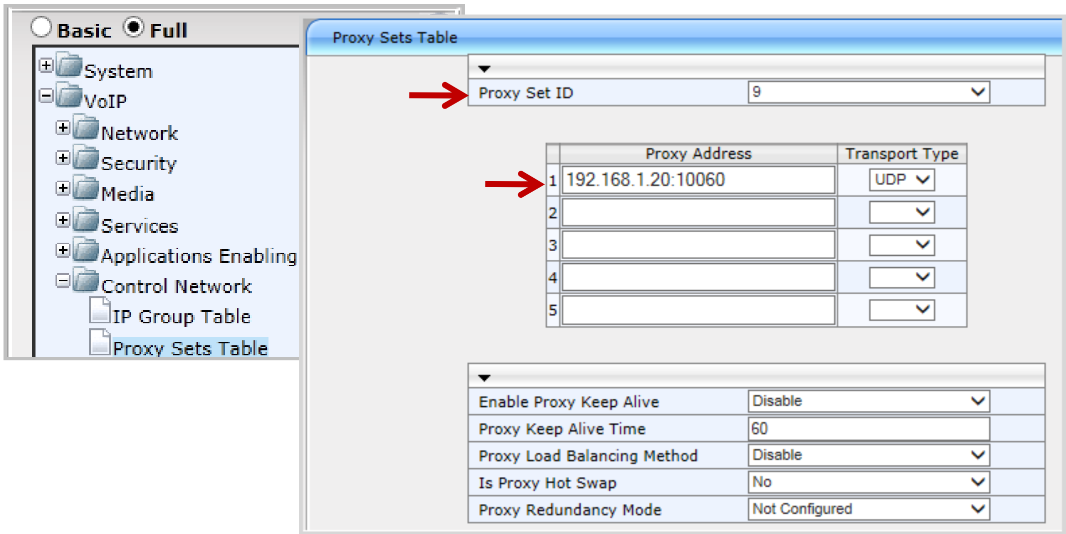


Figure 46: Configure the Proxy Sets table

2. Choose **Proxy Set ID 9**.
3. For entry 1, set the following parameters:
   - Set *Proxy Address* to **192.168.1.20:10060**
   - Set *Transport Type* to **UDP**
4. Now that you have enabled SAS, a new item appears on the menu. Choose **SAS** > **Stand Alone Survivability** and set the following parameters:

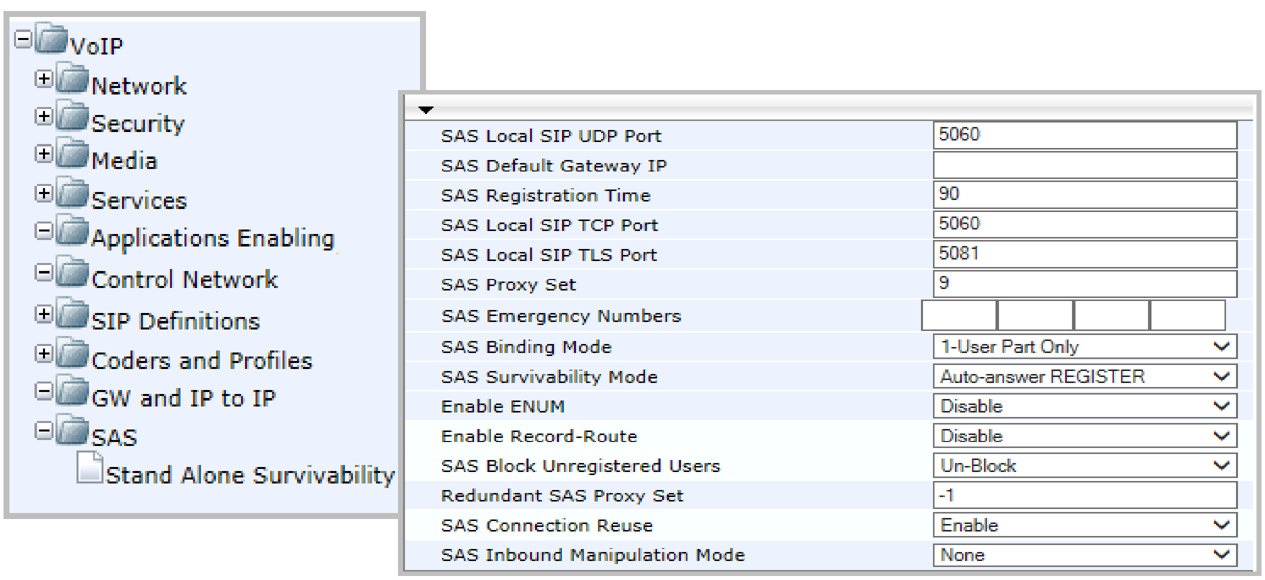
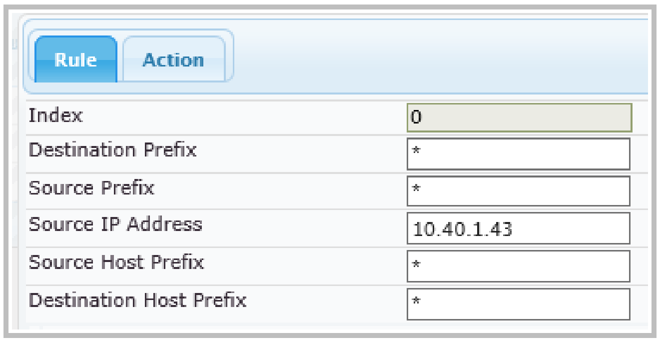


Figure 47: Set SAS parameters

- Set *SAS Local SIP UDP Port* to **5060**

- Set *SAS Registration Time* to **90**
- Set *SAS Proxy Set* to **9**
- Set *SAS Binding Mode* to **1-User Part Only**
- Set *SAS Survivability Mode* to **Auto-answer REGISTER**

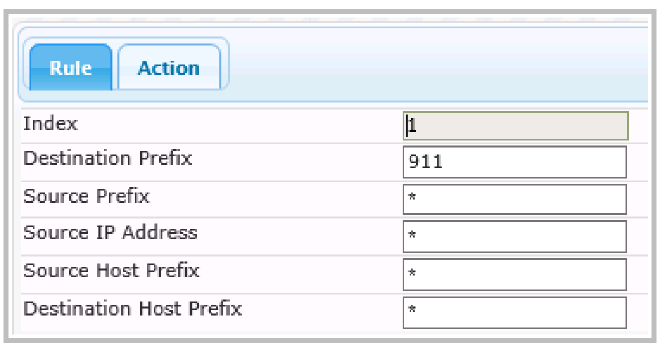
5. Select **VoIP** > **GW and IP to IP** > **Manipulations** > **Dest Number IP -> Tel**. You will add three new rules.
6. Click **Add** to add the first rule. Set the parameters on the *Rule* tab as shown in the next figure. There are no parameters to set on the *Action* tab.



This specifies that when the device receives a call from MaxCS (10.40.1.43), there is no need to remove the prefix “9.”

Figure 48: Set destination number manipulation rule 1

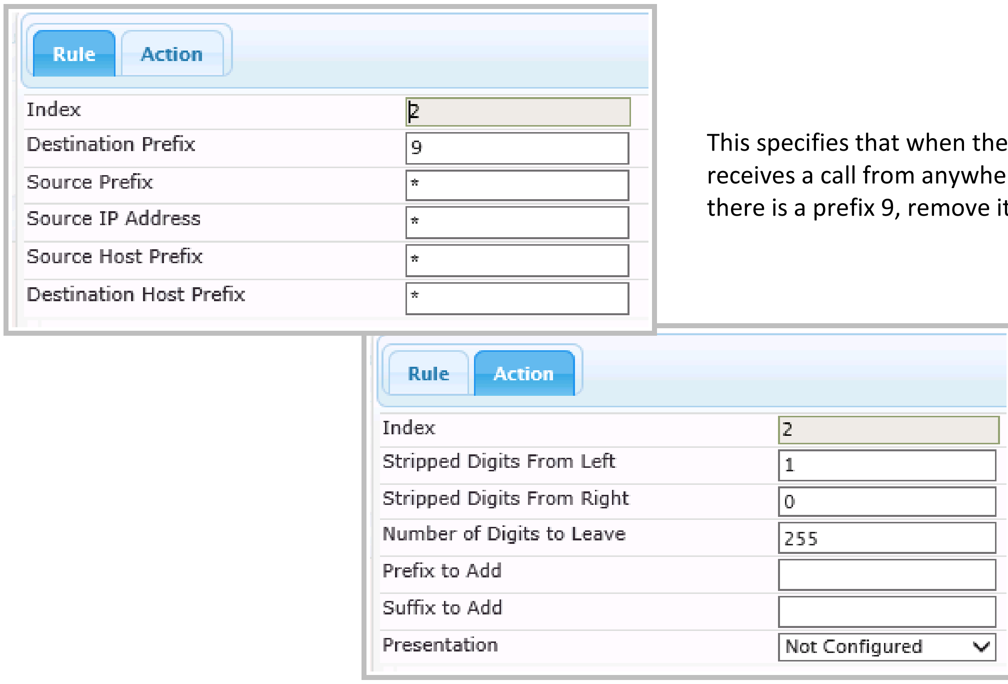
7. Click **Add** to add the second rule. Set the parameters on the *Rule* tab as shown in the next figure. There are no parameters to set on the *Action* tab.



This specifies that when the gateway receives a 911 call, it will not perform any digit manipulation.

Figure 49: Set destination number manipulation rule 2

8. Click **Add** to add the third rule. Set the parameters on both tabs, as shown in the next figure.



This specifies that when the gateway receives a call from anywhere else, if there is a prefix 9, remove it.

Figure 50: Set destination number manipulation rule 3

9. Select **VoIP** > **GW and IP to IP** > **Routing** > **IP to Hunt Group Routing**. Configure the rows as follows (note that row order is important for routing).

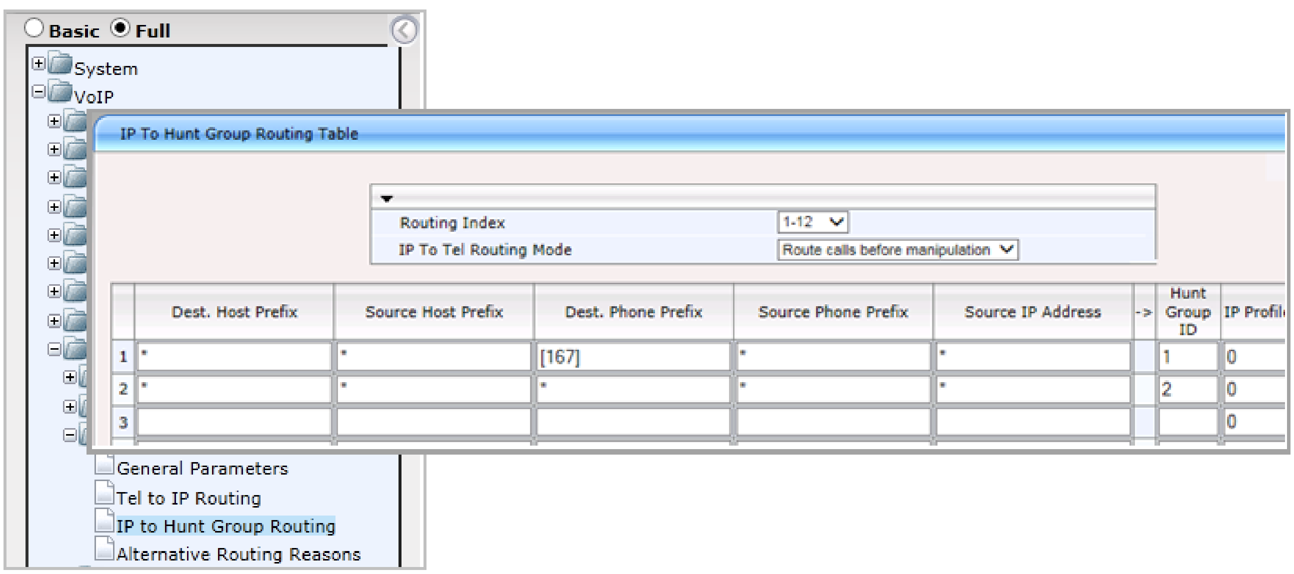


Figure 51: Configure IP to Hunt Group routing rules

a) For entry 1, if you already have an existing FXS IP to Hung Group Routing for extension 167, then set the *Source IP Address* to an asterisk (*****). The other fields remain the same.

When there is no survivable configuration, SIP messages can only come from MaxCS (10.40.1.43). However, when the device also acts as survival gateway, SIP messages can also come from different IP addresses. This is why we are placing the wildcard symbol (*) in this field.

IMPORTANT! Because the wildcard character is being used, be aware that any SIP device routable to the device could use the device to place PSTN toll calls. To prevent fraud calls, make sure your MP-11x device is behind a firewall. Also, no SIP port forwarding to the MP-11x device should be configured on the firewall, or else malicious users from the internet can use the device to place PSTN calls.

b) For other calls, route the call to FXO ports (Huntgroup ID 2). If there is an existing entry for this, change the *Source IP Address* to an asterisk (*****). if there is no such entry, add an entry 2 as follows:

- Set *Destination Host Prefix, Source Host Prefix, Destination Phone Prefix, Source Phone Prefix,* and *Source IP Address* to an asterisk (*).
- Set *Hunt Group ID* to **2**.

At this point, the general configuration is complete.

If you have a LAN or VPN-based deployment, review the architecture described in the next section before proceeding to the section *SAS IP Phone Configuration* on page 41.

If you have an Internet-based or Cloud deployment, skip ahead to the section *SAS FXO Configuration for Cloud Deployment* starting on page 36.

SAS LAN Deployment Architecture

This section shows the architecture for PSTN Survival FXO configuration on a local area network. This architecture also works for VPN-based deployments.

When the device loses the connection with the MaxCS server, incoming PSTN calls will loop back to the SAS service and route to extension 167. Calls from extension 167 will be routed to an FXO or FXS port, depending upon the digits that the caller dialed.

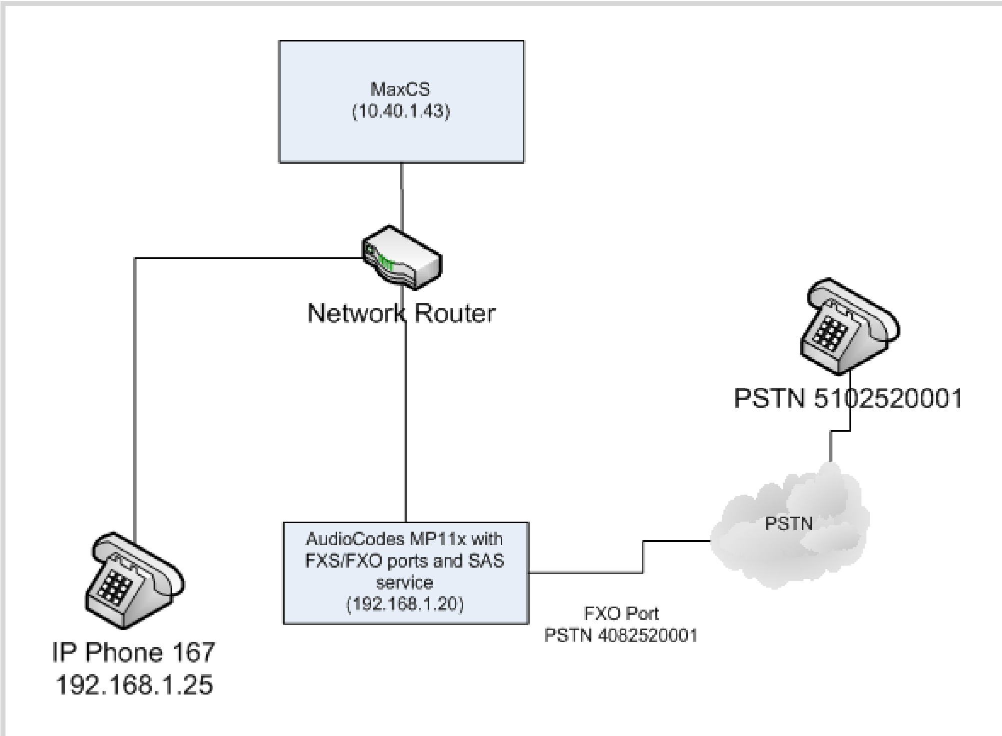


Figure 52: Architecture for LAN environment

You have completed the steps for LAN configuration; skip over the next section and continue with the instructions for configuring IP phones for PSTN survivability, which begin on page 41.

SAS FXO Configuration for Cloud Deployments

This section describes how to configure PTSN SAS for an internet or Cloud deployment.

Important! FXO port on an AudioCodes MP-11x device cannot talk to the MaxCS server SIP Trunk when the MP-11x device is configured behind a Firewall/NAT. However, when the MaxCS server is not reachable, IP phones and FXS ports on the MP-11x can register to the device's SAS service and use that service to make and receive calls. When the MaxCS server become available again, the IP phones and FXS ports will re-register back to the MaxCS server.

Cloud Deployment Architecture

The following diagram illustrates the architecture recommended for Internet/Cloud deployments.

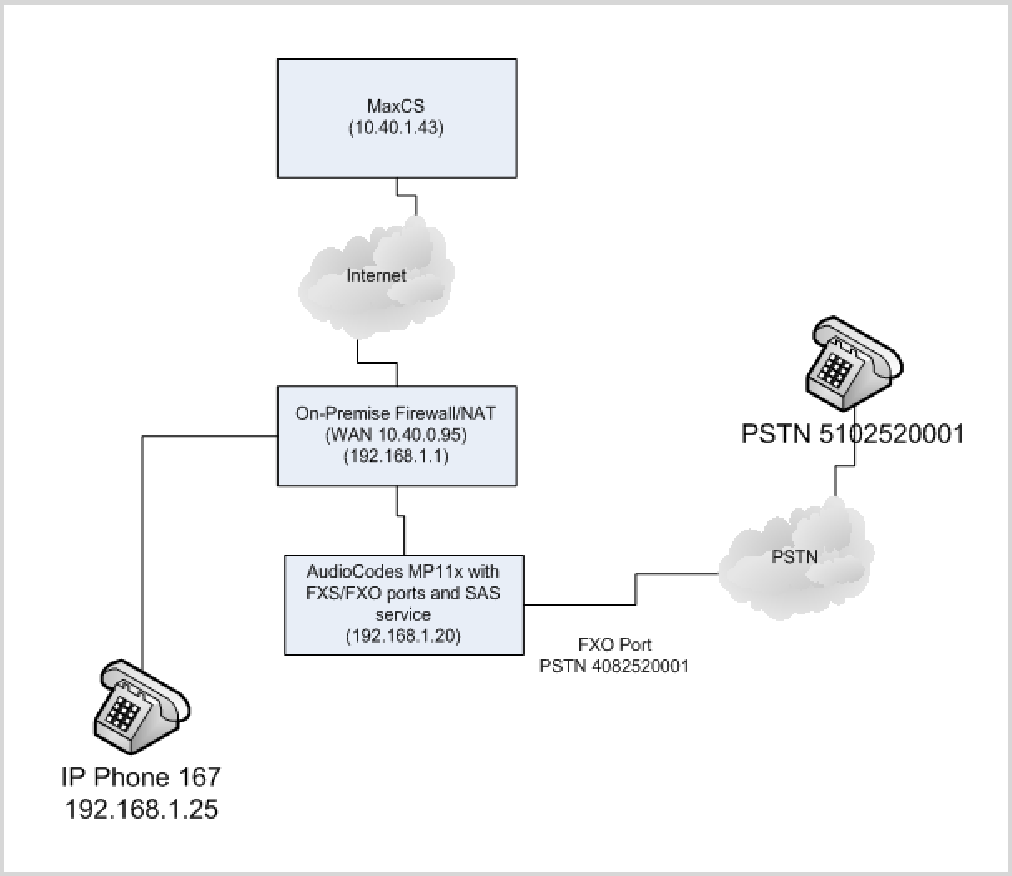


Figure 53: Architecture for Cloud environment

In our example in figure x, when a PSTN call comes into the FXO port but MaxCS is not reachable, the call will be routed to extension 167. When a PSTN call comes into the FXO port and MaxCS is working normally, the call could be dropped, because extension 167 is registered to MaxCS (10.40.1.43) and it could ignore the incoming call from the FXO port. To avoid this issue, do not publish that PSTN number (4082520001). This FXO number should be for out-going only trunks.

Requirements

Your environment must meet the following requirements:

- The server must be running MaxCS Release 7.5.0.60x or later.

Prerequisites

Perform these steps before you proceed:

- Complete the steps in the section *PSTN SAS General Configuration* starting on page 31.
- If you are configuring only FXO, then you may omit the entries in the Digit Mapping field as mentioned on page [TBD].

- If you are using FXS ports, also complete the steps in the section *Analog Extension (FXS) Configuration* starting on page 13.

Configuration Procedures

Follow these steps to configure PSTN survivability for a Cloud environment.

1. Log into the AudioCodes configuration tool. Click **Configuration** in the top left corner and set the menu mode to **Full**.
2. Select **VoIP** > **Control Network** > **IP Group Table**.

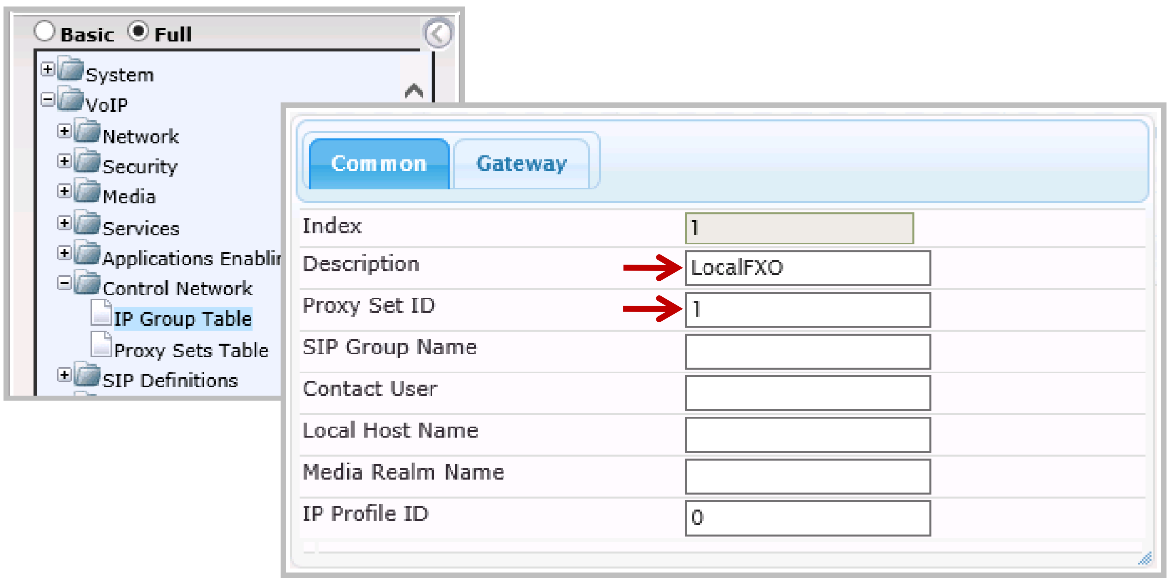


Figure 54: IP Group table parameters

3. On the *Common* tab, set the following parameters:
   - Set *Description* to **LocalFXO**
   - Set *Proxy Set ID* to **1**

4. Select **VoIP** > **Control Network** > **Proxy Sets Table**. For entry 1, set the following parameters:

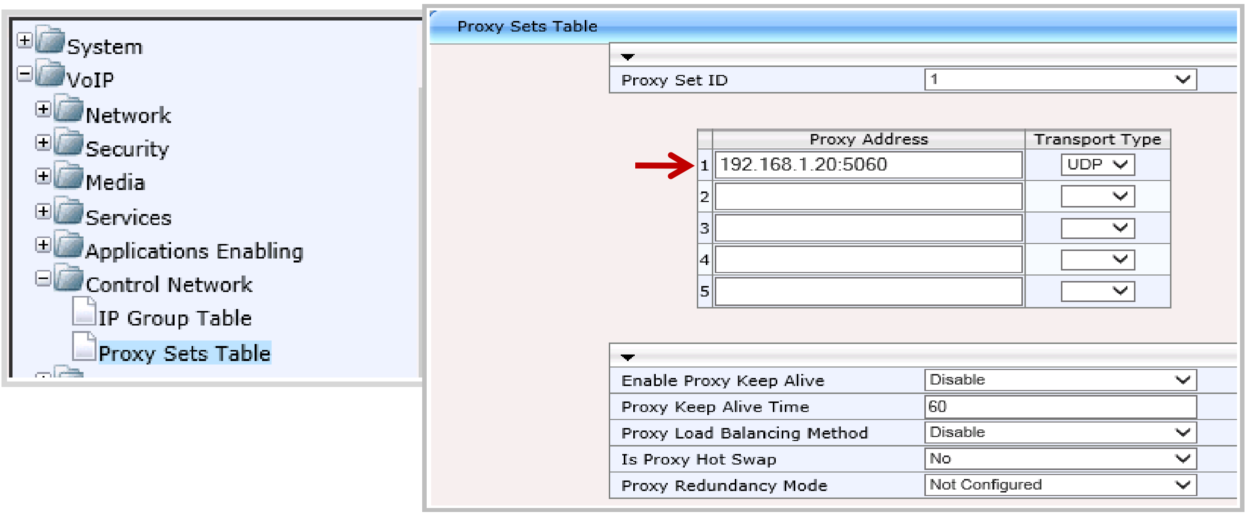


Figure 55: Proxy Sets table parameters

- Set *Proxy Address* to **192.168.1.20:5060**
- Set *Transport Type* to **UDP**

This proxy set will be used later for FXO ports (Huntgroup ID 2). When a PSTN call comes in to 4082520001 (refer to Figure 53), it will not be sent to MaxCS at 10.40.1.43 because MaxCS cannot [TBD – access?] the MP-11x FOX port behind NAT. The call will be sent directly to the SAS service in the MP-11x device (192.168.1.20:5060).

5. Select **VoIP** > **GW and IP to IP** > **Hunt Group** > **End Point Phone Number**. In addition to any extension numbers that may have already been configured, add trunk numbers. Note that empty rows between the FXS and FXO channels do not matter.

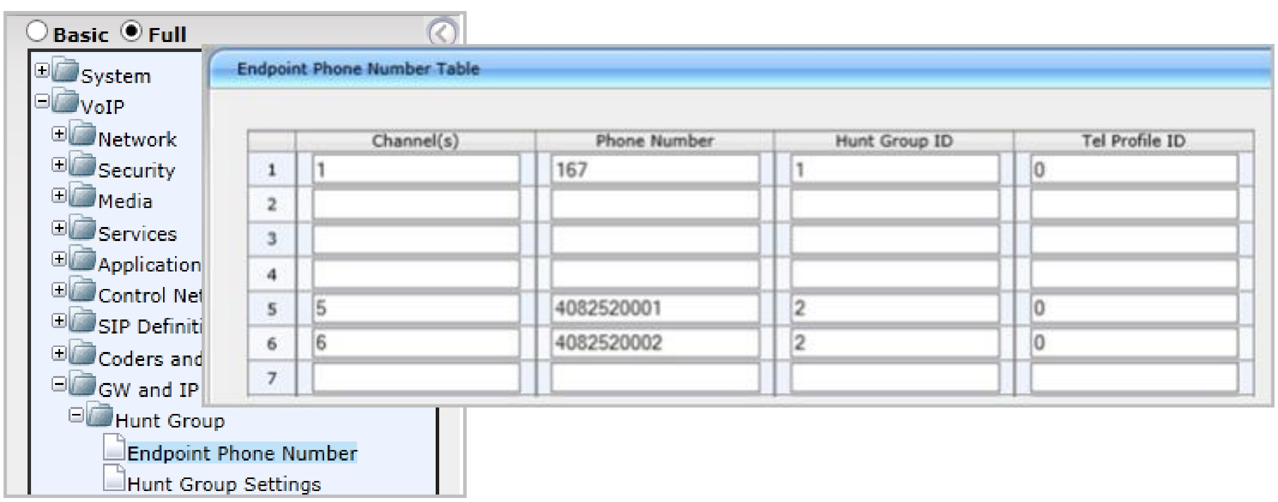


Figure 56: Endpoint Phone Number table

Set the following parameters:

- Set entry 5 to channel **5**. (On the MP-118, this is the first FXO port.)
- For channel 5, set the *Phone Number* field to your PSTN phone number (in our example, this is 4082520001) and set *Hunt Group ID* to **2** (FXO group).

Entry 5 indicates that for incoming calls to the FXO port, if a caller ID is not received, the number 4082520001 will be used as the default caller ID. The gateway will forward the call to MaxCS with the caller ID through SIP.

6. Add more channels as needed; in Figure 56, channel 6 has also been configured.

7. Select **VoIP** > **GW and IP to IP** > **Hunt Group** > **Hunt Group Settings**. Configure a hunt group ID 2 as follows.

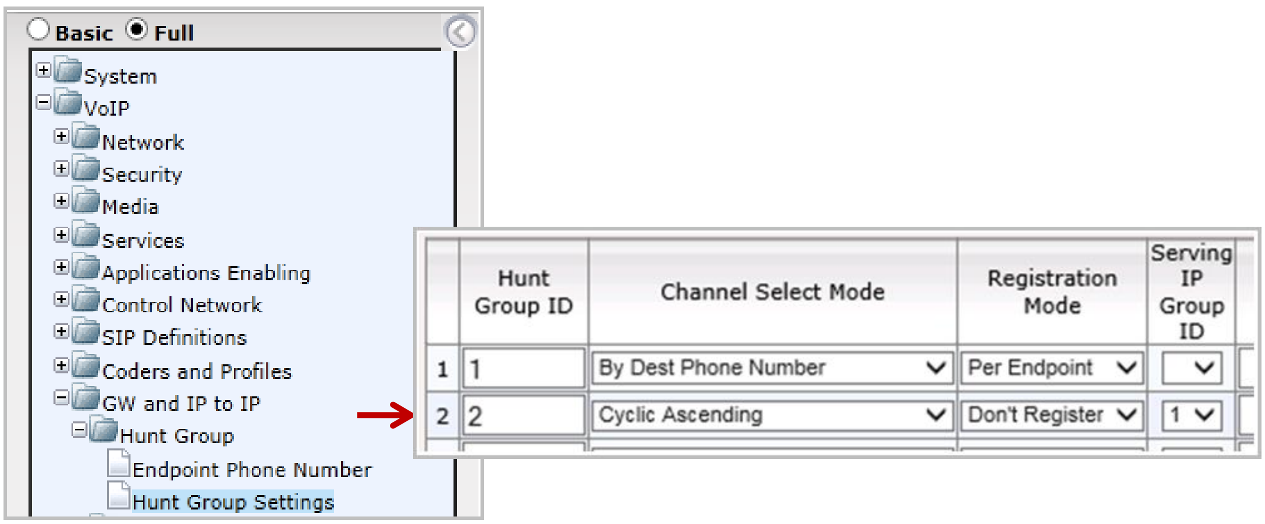


Figure 57: Hunt Group 2

- Set entry 2 to Hunt Group ID **2**.
- Set *Channel Select Mode* to **Cyclic Ascending**.
- Set *Registration Mode* to **Don’t Register**.
- Set *Serving IP Group ID* to **1**.

8. Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **FXO Setting**. Set *Dialing Mode* to **One Stage**; set *Discon-nect on Dial Tone* to **Enable**.

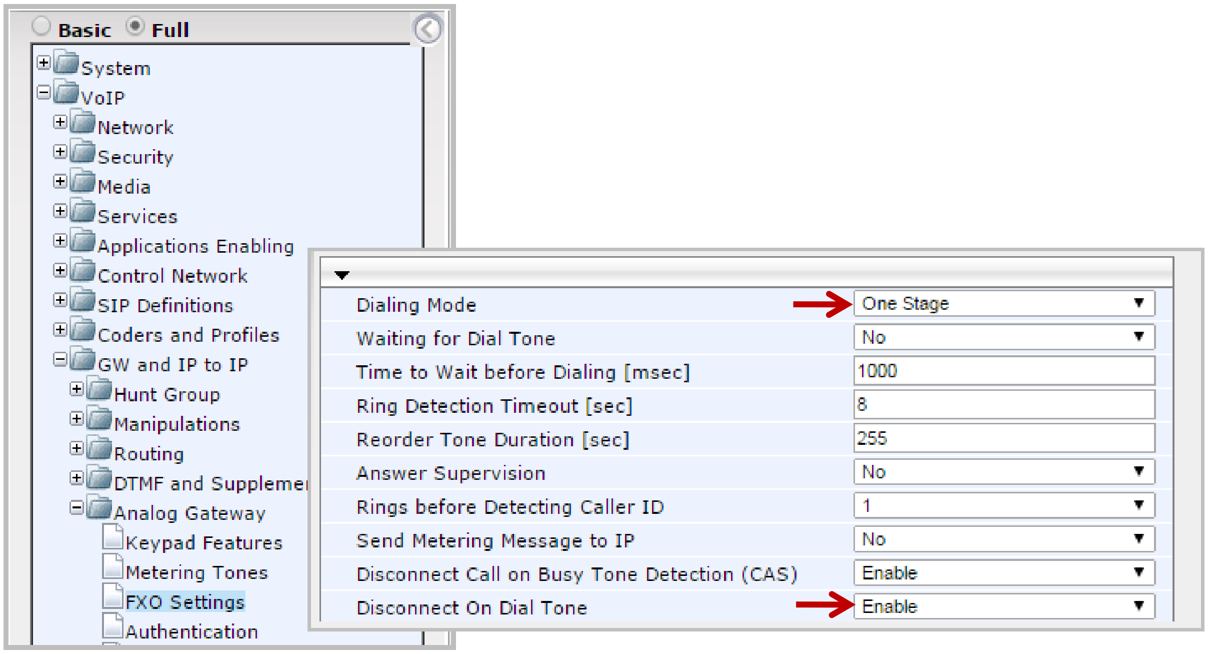


Figure 58: Configure FXO parameters

Configuration for Cloud and Internet environments is now complete. Continue with the next section to configure IP phones for PSTN Survival mode.

SAS IP Phone Configuration

This section describes how to configure IP Phones for PSTN survivability.

When MaxCS is not reachable, extension 167 can send the call to MP-11x (192.168.1.20). If registration is enabled on extension 167, it can also receive calls from MP-11x (1992.168.1.20).

Note: An MP-11x can only support up to 25 con-current SIP registration, so if you have more than 25 extensions in a system, enable the SIP registration only on the extension that is required to receive incoming call when MaxCS is not reachable.

There are 3 kinds of IP phones that are supported. This section includes the procedures for all three types.

- AltiGen IP phones (705, 710, 720, and 805)
- Polycom IP phones
- Extensions attached to an MP-11x's FXS port

Prerequisites

- If you have a LAN-based deployment, complete the steps in the section *PSTN SAS General Configuration* and review the architecture on page 35 before perform the steps in this section.

  If you have an Internet/Cloud based deployment, complete the steps in the section *SAS FXO Configuration for Cloud Deployments* before start this section.

- Before you perform the steps in this section, confirm that your extension (in our example, this is extension 167) can make send and receive calls when MaxCS is reachable.

Follow this process to configure IP phones or the FXS port acting as an IP extension:

1. Select **VoIP** > **GW and IP to IP** > **Analog Gateway** > **Automatic Dialing**. Confirm that the entry is set to the extension number (in our example, extension 167), so that when a PSTN call comes in, it will ring that IP extension.

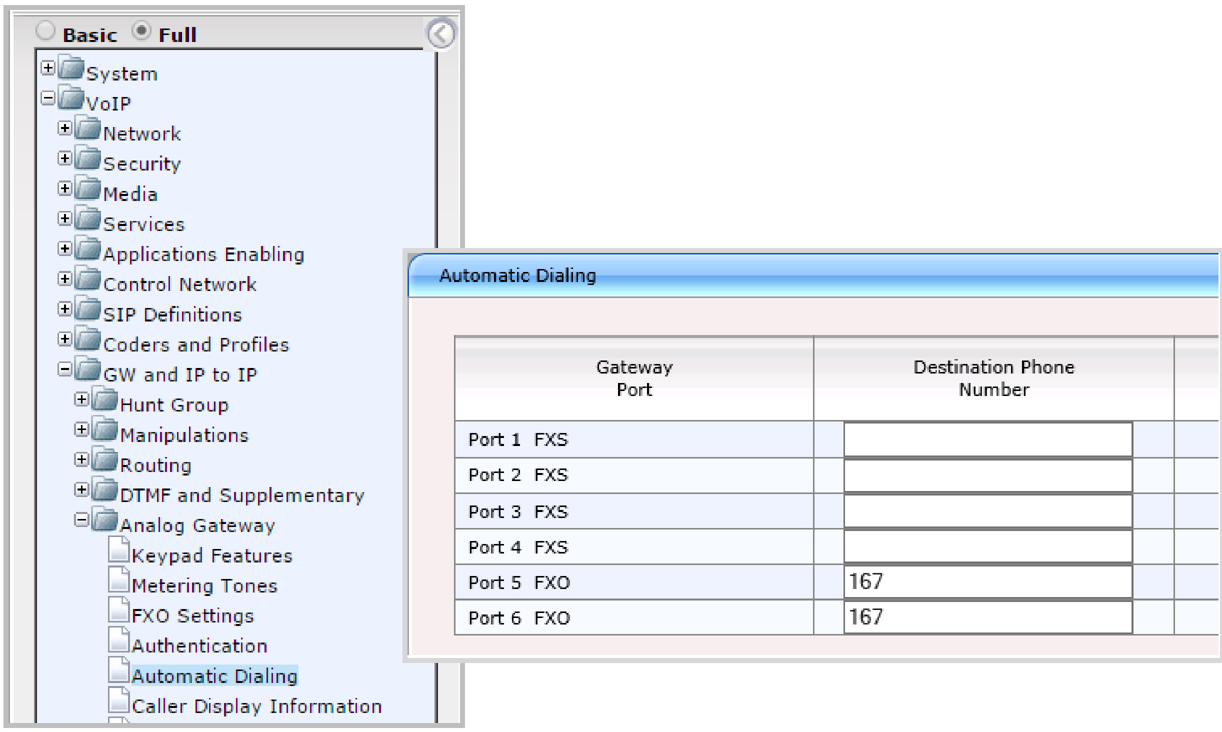


Figure 59: Set Automatic dialing parameters

2. For AltiGen IP phones IP705,710, 720, and 805:
   a) Open the phone's menus. Select **System** > **Emergency GW** and set the *IP address* to **192.168.1.20**.
   b) Then set **System** > **Enable SIP Registration** to **Yes** only if this phone needs to receive incoming calls while MaxCS is not reachable. Otherwise, set this to **No**.

3. For Polycom IP phones:
   a) Open MaxCS Administrator.
   b) Select **PBX** > **AltiGen IP Phone Configuration**. Switch to the *Polycom* tab. (This tab is available starting with MaxCS Release 7.5.0.603.)
   c) Choose the extension. Check the option *Enable Secondary Proxy*. Set the *IP Address* to **192.168.1.20**. Set the *Port Number* to **5060**.
   d) Check *Enable Registration* **only** if the phone needs to receive incoming calls while MaxCS is not reachable. Otherwise, leave this option cleared. Note that if a Polycom phone is used for this ex-

tension, in order for the user to use #17 on the phone, the extension must not have *Enable Registration* checked.

In some cases, Polycom phones will need to be rebooted twice before the phone fully incorporates the changes.

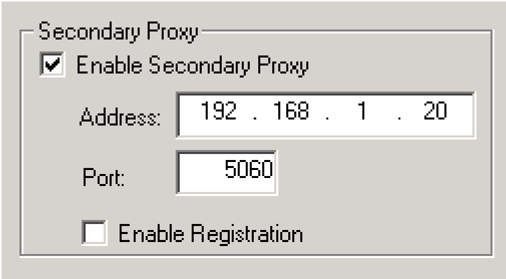


Figure 60: Polycom Secondary Proxy settings

4. For the MP-11x FXS port acting as an IP extension:

   a) In the gateway web configuration tool, select **VoIP** > **Control Network** > **Proxy Sets Table**.

   b) Configure Proxy Set ID **0**:

      a. *Proxy Address* entry 1 should already have been configured, because your extension on the FXS port was already working while MaxCS is reachable.

      b. For *Proxy Address* entry 2, enter **192.168.1.20:5060**.

      c. Set *Enable Proxy Keep Alive* to **Using Options**.

      d. Set *Proxy Keep Alive Time* to **15**.

      e. Set *Proxy Redundancy Mode* to **Homing**.

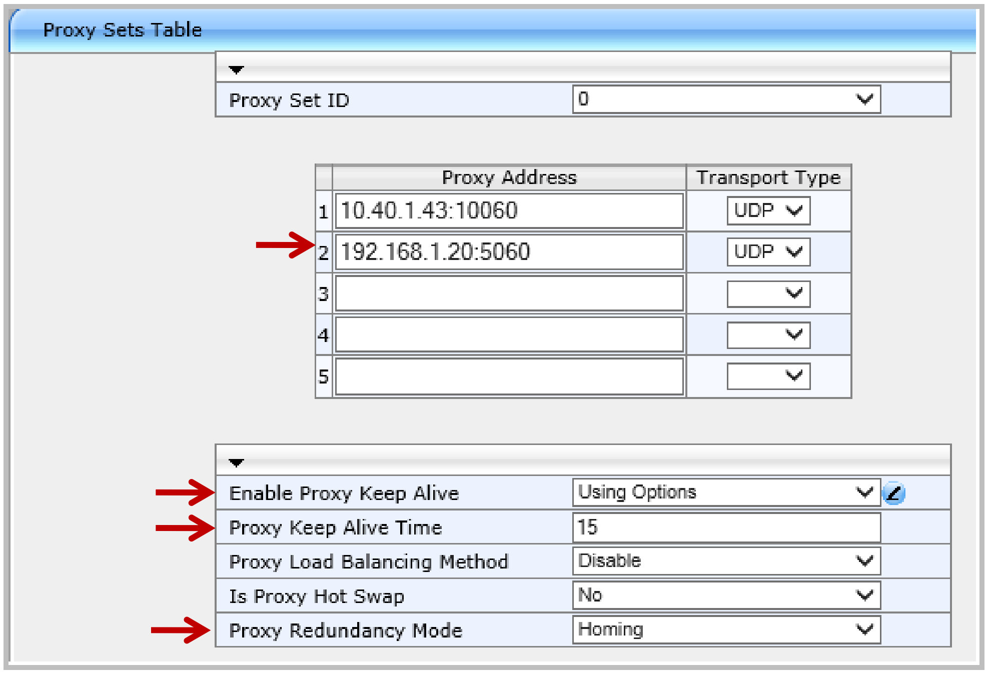


Figure 61: Proxy Set ID 0 parameters

c) Submit the change and click **Burn**.

d) You must modify the registry. On the MaxCS server, run *regedit* as an administrator.

e) Export your current registry to a file, as a backup. Be sure to follow Microsoft's recommended procedures for updating your registry.

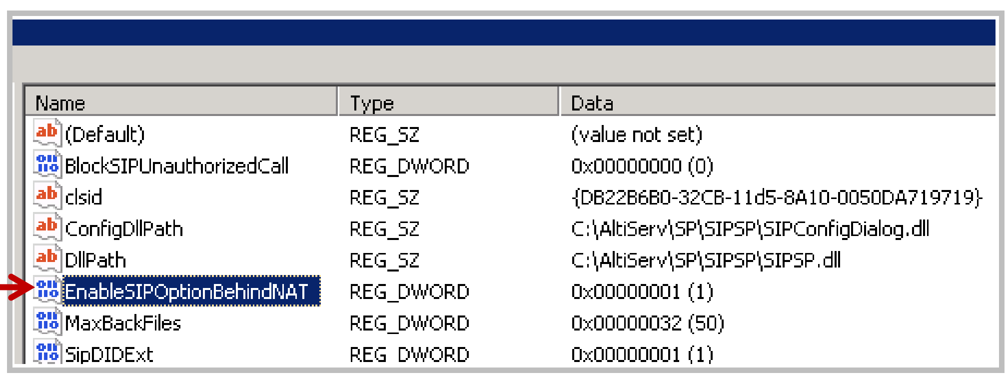


Figure 62: Update the registry

f) In the registry, change the EnableSIPOptionBehindNAT key to a value of **1**.

On 64-bit Windows systems, this key is located here:

HKEY_LOCAL_MACHINE\SOFTWARE\Wow6432Node\AltiGen Communications, Inc.\AltiWare\Service Providers\SIPSP\

On 32-bit Windows systems, this key is located here:

HKEY_LOCAL_MACHINE\SOFTWARE\AltiGen Communications, Inc.\AltiWare\Service Provid-ers\SIPSP\

g) Save these changes.

h) Restart the MaxCS services (or reboot the MaxCS server).

Confirm the Configuration

Follow these steps to confirm that your system is configured correctly.

1. Confirm that when the MaxCS server is reachable, you can make and receive calls through IP extension 167.

2. Shut down MaxCS.

   At this point, on most AltiGen IP phones, the status display will change from ‘basic’ to ‘local.’

   For Polycom phones, if *Enable Registration* was not checked in Max Administrator, the phone will not be able to register and will show a red ***x*** beside the extension number.

   Even on phones that are not registered, users can still go off-hook and dial a PSTN number or an exten-sion number.

3. From the phone, dial 915102520001 and press #. The call should connect properly.

4. If you enabled registration for the phone, log into the MP-11x web configuration tool and click **Status and Diagnostics** above the menus. Then select **VoIP Status** > **SAS/SBC Registered Users**. You should see extension 167 in the *Address of Record* field.

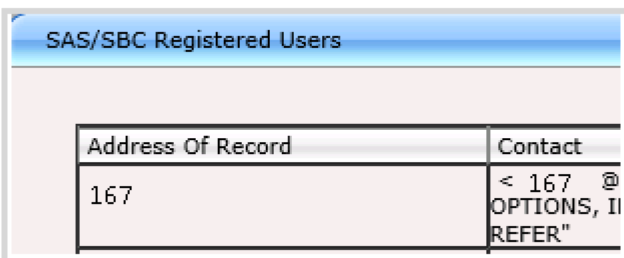


Figure 63: Registered users shows extension 167

5. Make a call to 14082522001 (this is the FXO port’s PSTN number) from your mobile phone. IP phone 167 should ring. Pick up the call and verify the connection.

6. Restart the MaxCS server. IP phone 167 should reconnect to the MaxCS server shortly. The length of the delay varies greatly, depending upon the size of the system; on small systems this may take 2-3 minutes; on large MaxCS systems it may take up to 10 minutes.

7. Configure another IP Phone with extension 168 with registration turned on. Shut down the MaxCS serv-er again and check that the extension is registered (just as you did in step d. Make a call from IP phone 168 to IP phone 167; answer the call and verify the connection.

Centrex Configuration (Optional)

This section applies to organizations that have PSTN Centrex or have FXO ports connected to analog extensions on a PBX such as Nortel.

This configuration allows the system to release the FXO port after a transfer. For example, FXO port 1 receives a call. The call is transferred. AudioCodes sends a Centrex flash to the same FXO port to complete the transfer. Once the transfer is successful, the FXO port is freed up to accept another call.

When FXO Centrex is enabled, MAXCS will not support FXS ports on the gateway because SIP INFO must be turned off. In other words, you may consider the MP-114 and MP-118 all-FXO ports models, because FXS ports on the gateway will not be usable.

These instructions apply only to MaxCS release 7.5.0.60x and later.

Before you attach the trunk to the AudioCodes gateway, make sure that your trunk supports Centrex transfer. To confirm this support, attach an analog extension to a CO Centrex line and perform some Centrex transfer from the analog extension manually.

Before you begin these procedures, perform all of the steps in the section FXO Configuration starting on page 21. Then confirm that you can make and receive calls through an FXO port without issues.

1. In your browser, navigate to this URL:

   http://192.168.1.20/AdminPage

   Substitute the gateway's IP address for 192.168.1.20.

2. On the left, select **ini Parameters**.
3. For the parameter *LINETRANSFERMODE*, enter **3**. Click **Apply New Value**.
4. For the parameter *DTMFDIGITLENGTH*, enter **300**. Click **Apply New Value**.

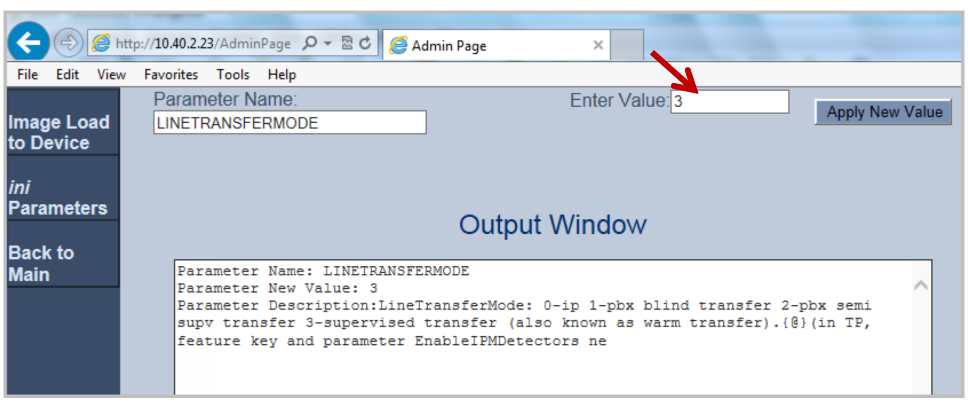


Figure 64: Specify INI parameters

5. Log into the AudioCodes web configuration tool, and set the menu mode to **Full**.

6. Select **VoIP** > **Media** > **IPMedia Settings**. Set *Answer Detector Activity Delay* to **300**.

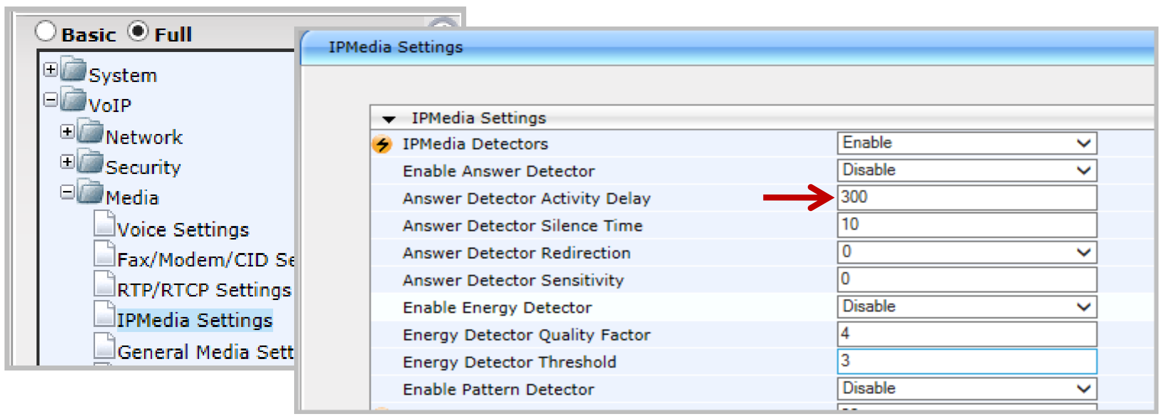


Figure 65: Adjust IP Media parameter

7. Select **VoIP** > **GW and IP to IP** > **DTMF and Supplementary** > **Supplementary Services**. Set *Enable Hold* to **Enable**; set *Enable Transfer* to **Enable**.

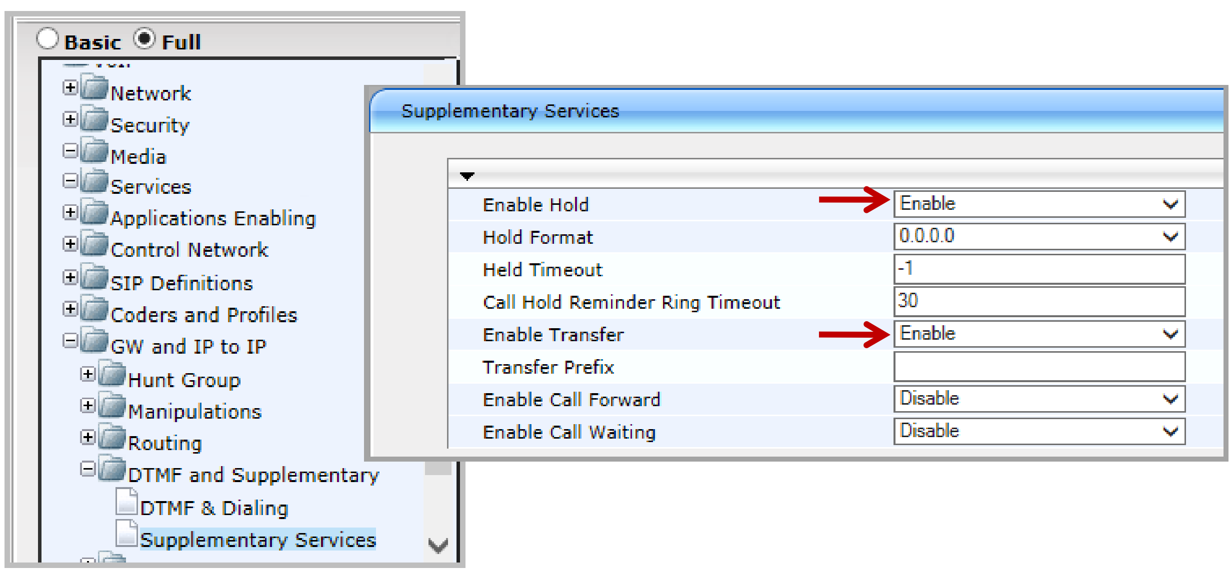


Figure 66: Adjust Supplementary Services parameters

8. Log into MAXCS Administrator and double-click **SIPSP** in *Boards* view. Click **SIP Trunk Configuration.** Click **SIP Trunk Profile**.

   Create a new SIP Trunk profile named *centrex*. For that profile, check *Enable SIP REFER* and *Enable Centrex Transfer*.

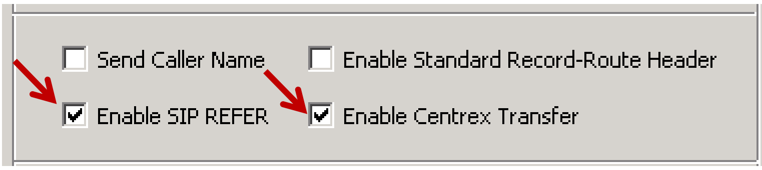


Figure 67: Create a new SIP Trunk Profile

9. Close the open panels. In *Boards* view, double-click **SIPSP**. Click **Board Configuration** and then click **SIP Trunk Configuration**.
10. Set *SIP Trunk Profile* to **centrex**. Apply this setting to the other SIP trunks where SIP Sever IP addresses are 192.168.1.20.

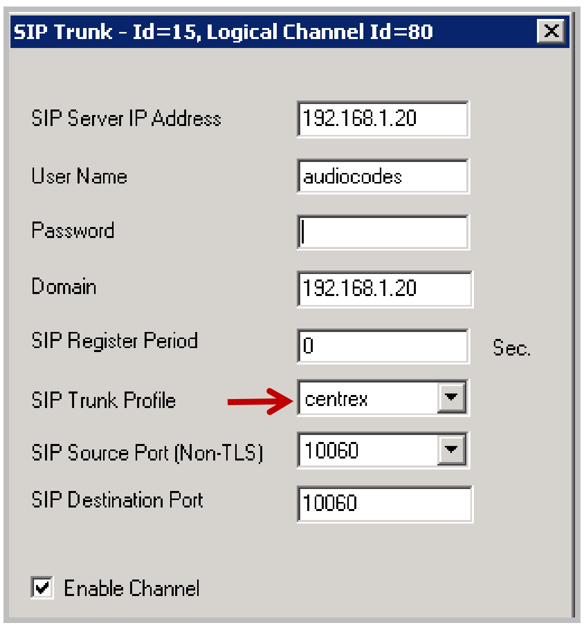


Figure 68: Set SIP Trunk Profile to centrex

11. Select **PBX** > **Extension Configuration**. Create an extension 150, and for that extension, on the *General* tab, check *Release SIP Tie-Link Trunk*.

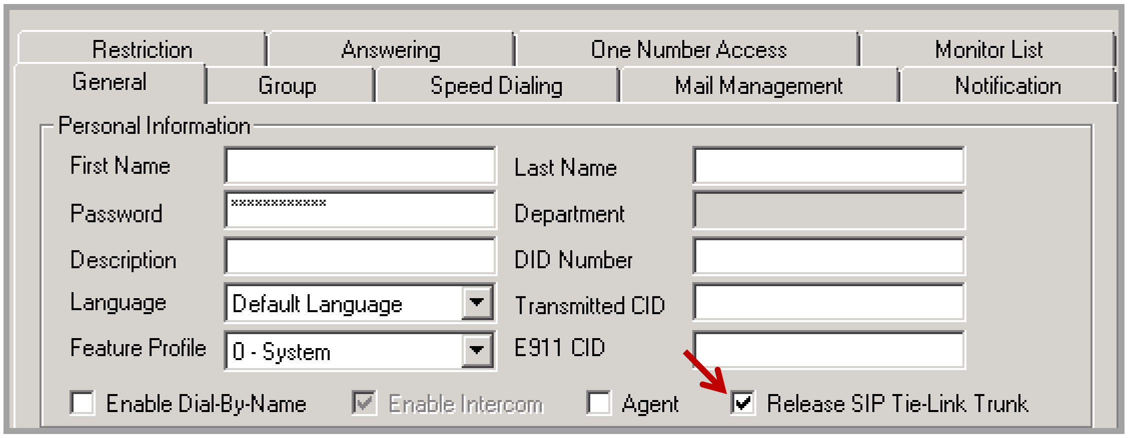


Figure 69: Configure extension 150

12. Switch to the *Restriction* tab. Under the *Other Call Restrictions* section, check the first two options, *Allow Calls to be Transferred or Conference to an Outside Number* and the *Allow Extension User to Configure Forwarding, Notification and Reminder Call to an Outside Number* options.

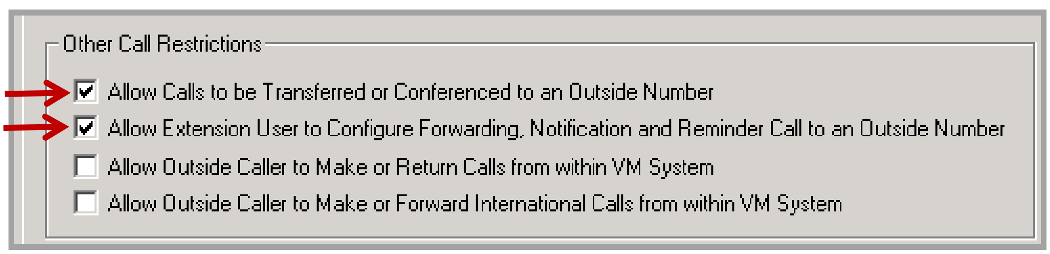


Figure 70: Set call restriction options for extension 150

13. Switch to the *Answering* tab. Check *Enable Forward to* and set it to **Free Format**.

    Set the number to the PSTN number with the trunk access code. In our example, it is 915102520001,,,,,.

    The commas set a delay before MAXCS releases the centrex line to finish the transfer. Each comma inserts a one-second delay after the call is forwarded. Use at least five commas (for five seconds). Longer numbers may require additional commas. However, too many commas will impact the cut through time.

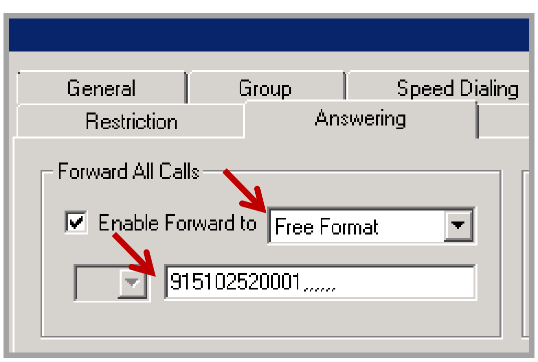


Figure 71: Set forwarding to Free Format

14. Follow these steps to verify that the configuration is correct:

    a) From a mobile phone, make a call to the AudioCodes FXO port. In this example, the number is 14082520001.

    b) Confirm that the call routes to the AltiGen's IVR system.

    c) In the IVR system, dial the virtual extension number you just created – extension 150.

    d) PSTN phone 15102520001 should ring. Answer the call and confirm that you can hear voice.

    e) The SIP trunk in MAXCS should be available.

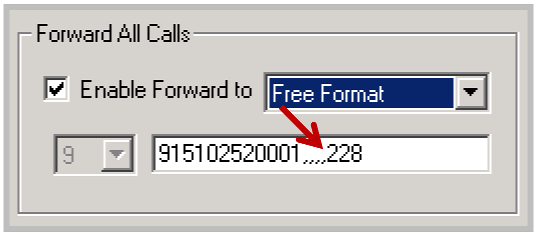


Figure 72: Inject numbers after the IVR system answers

You can use this feature to inject a number (for example, an extension number) after the IVR system answers.

Troubleshooting Tips

- Usually, the gateway's FXO port will connect to a PBX extension port (FXS port) or a CO Centrex line. Make sure that MWI (Message Waiting Indicator) is turned off for the FXS port or CO line (if it has a MWI). The MWI signal may be misconstrued as phantom calls.
- When performing a Centrex transfer to an invalid destination, sometimes the CO or PBX's FXS port will play error or busy tones. When the gateway detects the tone during this transfer, it may send another flash-hook to the CO or PBX FXS port. Usually this will not cause an issue.

TLS Configuration (Optional)

This section includes instructions for enabling TLS for FXS and FXO configurations.

Be aware of the following considerations before you configure TLS for your gateway:

- When TLS is enabled on MP-1xx devices, the total number of channels that are available is reduced. This reduction is the result of the additional processing power that is required for TLS. This limitation comes from AudioCodes.
    - MP-114 devices will be reduced from 4 channels to 3 channels.
    - MP-118 devices will be reduced from 8 channels to 6 channels.
    - MP-124 devices will be reduced from 24 channels to 18 channels.
- You cannot enable TLS on an MP-1xx device that includes both FXS and FXO channels. If you need both FXS and FXO channels with TLS enabled, you must use two different MP-1xx devices.

- You cannot enable both Survivability and TLS on the same MP-1xx device. Consider using AltiGen or Polycom phone to achieve survivability when TLS is also required.
- You cannot configure both Centrex and TLS on the same MP-1xx device.

Follow these steps to begin the configuration, and then continue with the FXO or FXS configuration sections.

1. Select **VoIP** > **Media** > **Media Security**. Set *Media Security* to **Enable**.

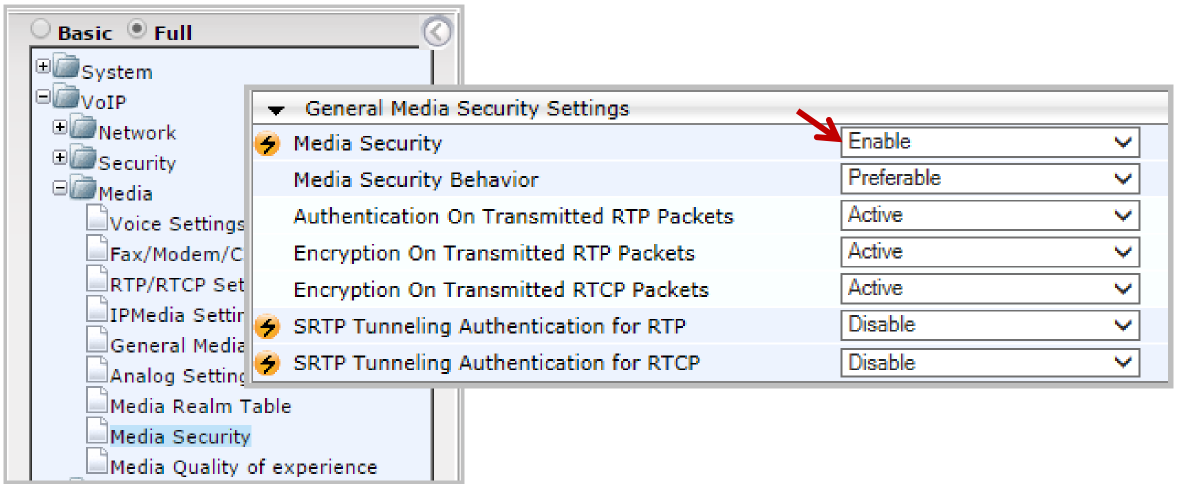


Figure 73: Enable media security

2. Submit the change and click **Burn**. Reboot the device.
3. After the gateway reboots, open the Home page. You will find that the number of channels has been reduced.

   The next figure illustrates an AudioCodes MP-114 gateway whose channels have been reduced from 4 channels to 3 channels.

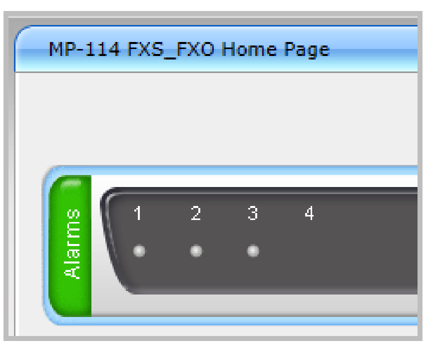


Figure 74: Example of an MP-114 with reduced to only 3 channels

4. Select **VoIP** > **Control Network** > **Proxy Sets Table**. Update *Proxy Set ID 0* as follows:
   - Change *Proxy Address* to **10.40.1.43:5061**
   - Change *Transport Type* to **TLS**

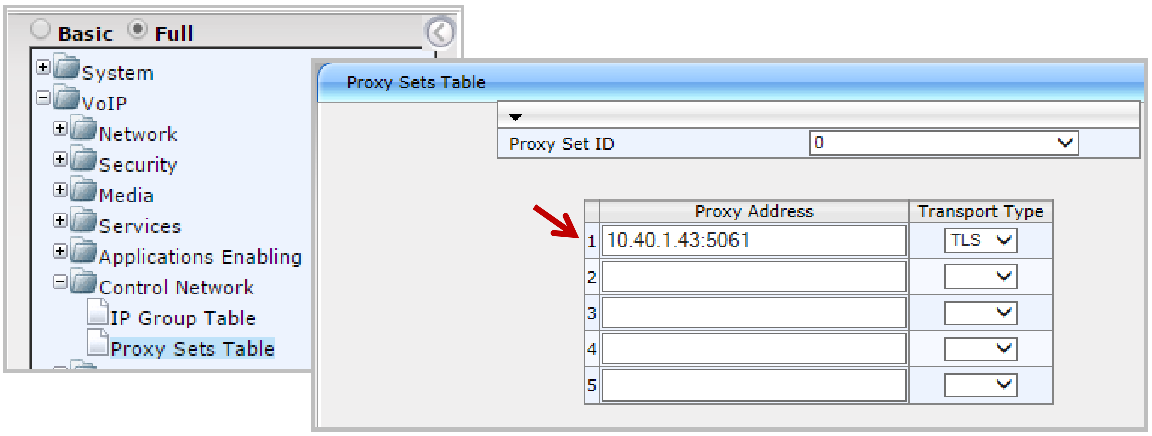


Figure 75: Edit proxy set 0

5. Submit the change and click **Burn**.

Continue with one of the next two sections as appropriate.

TLS Configuration for an FXS Port

Follow these steps to enable TLS for an extension. These instructions use extension 167 in the examples.

1. Complete all of the steps in the section *Analog Extension (FXS) Configuration* (page 13) to configure the extension without TLS. Confirm that the extension can place and receive calls, transfer calls, and join conferences before you proceed.
2. Log into MaxCS Administrator and select **PBX** > **AltiGen IP Phone Configuration**. Switch to the **General** tab.
3. Select extension 167 and check both of the two *SIP Transport* options, **Persistent TLS** and **SRTP**.

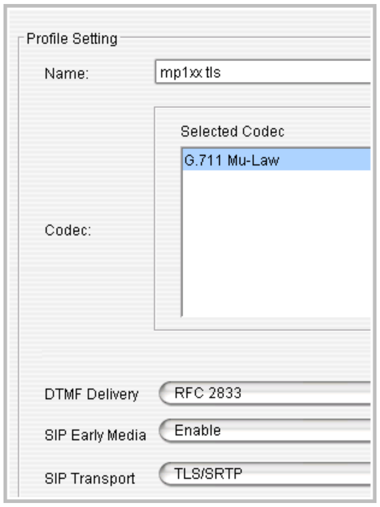
Figure 76: Check the SIP Transport options

4. Test whether extension 167 can make and receive calls.
5. If a Fax device is attached to this extension, perform the steps in the section *Analog Extension (FXS) Configuration* on page **Error! Bookmark not defined.**.

TLS Configuration for an FXO Port

Follow these steps to enable TLS for an FXO port.

1. Complete the steps in the section, FXO Configuration (page 21) to configure the FXO trunks without TLS. Make sure that you can use the trunk to make and receive calls before you proceed.
2. Log into MaxCS Administrator and select **VoIP** > **Enterprise Network Management**. On the top toolbar, click the **Codec** button.
3. Create a new codec profile named **MP1xx TLS**. Configure the following settings for this profile:



- Set *Selected Codec* to **G.711 Mu-Law**
- Set *DTMF Delivery* to **RFC 2833**
- Set *SIP Early Media* to **Enable**
- Set *SIP Transport* to **TLS/SRTP**

Figure 77: Configure the new codec profile

4. In Enterprise Manager, click the **Servers** button on the top toolbar. Select the **IP Codec** tab.
5. Find the entry with the MP-1xx's IP address. In the example in the next figure, the IP address is 192.168.1.20.

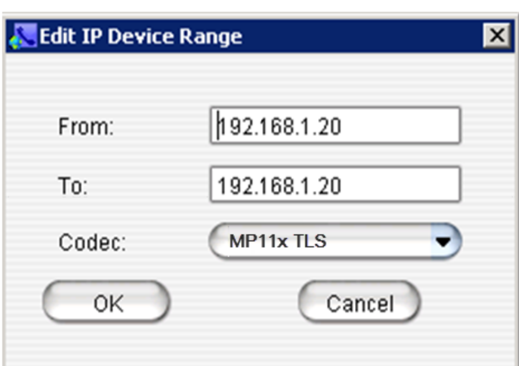


Figure 78: Select the codec profile

6. Set *Codec* to **MP1xx TLS** (the one you just created).
7. Open Boards view. Double-click **SIPSP**, click **Board Configuration**, and then click **SIP Group Configuration**.
8. Edit the SIP server that was configured to point to the MP-1xx device to change *SIP Destination Port* to **5061**.

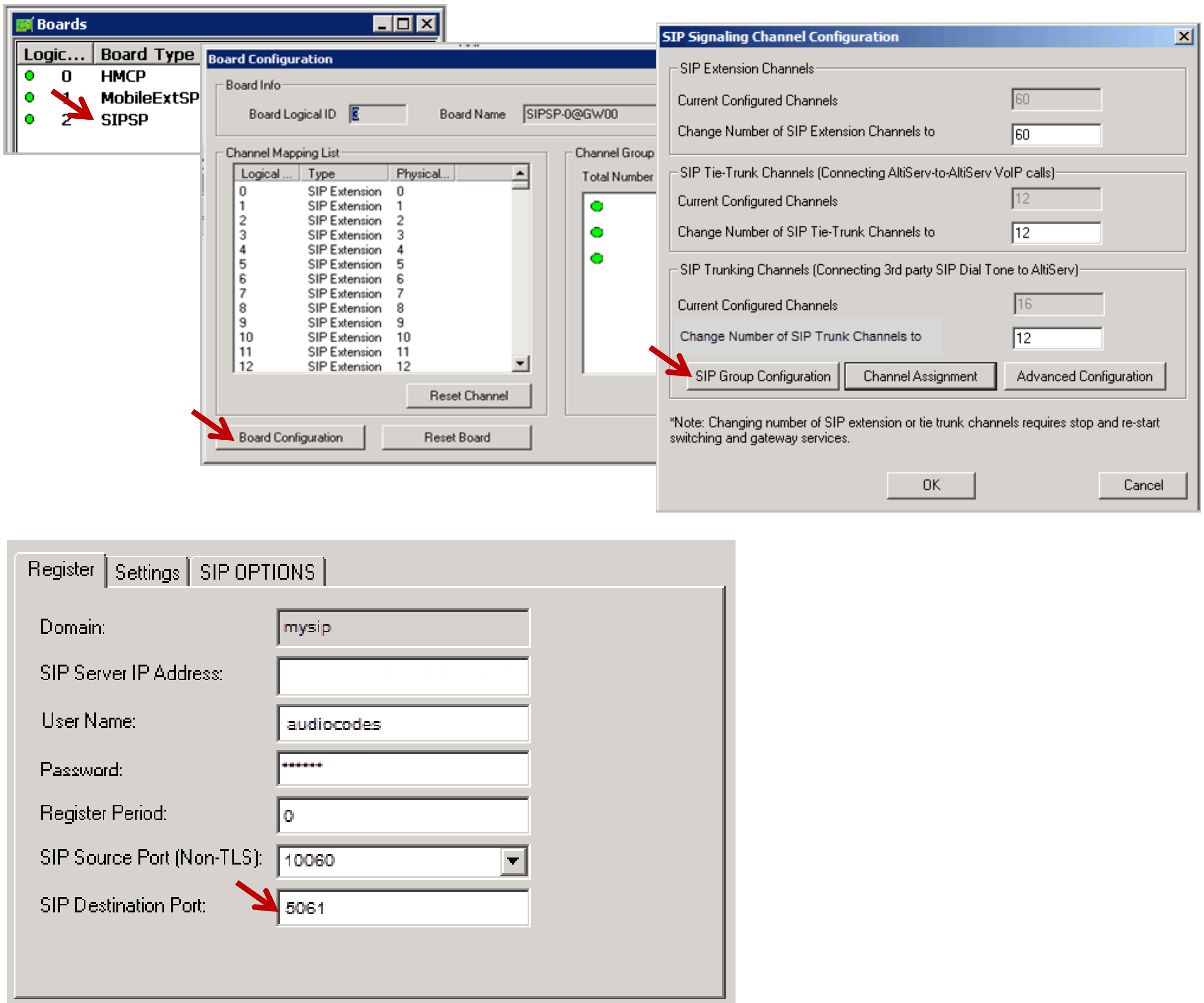


Figure 79: Update the SIP Destination port

9. Verify this setting by making outbound calls and receiving inbound calls using the FXO port.

Resetting the Gateway to Default Settings

We recommend that you reset the gateway to its original factory default settings before you begin your configuration.

Note that your current FXS extension password will be lost if you restore the current configuration. This is because the FXS extension password will not be stored in the .ini file. You will need to re-enter the FXS extension authentication password if you restore the .ini file. To do this, choose **VoIP** > **GW and IP to IP** > **Analog Gateway** > **Authentication**.

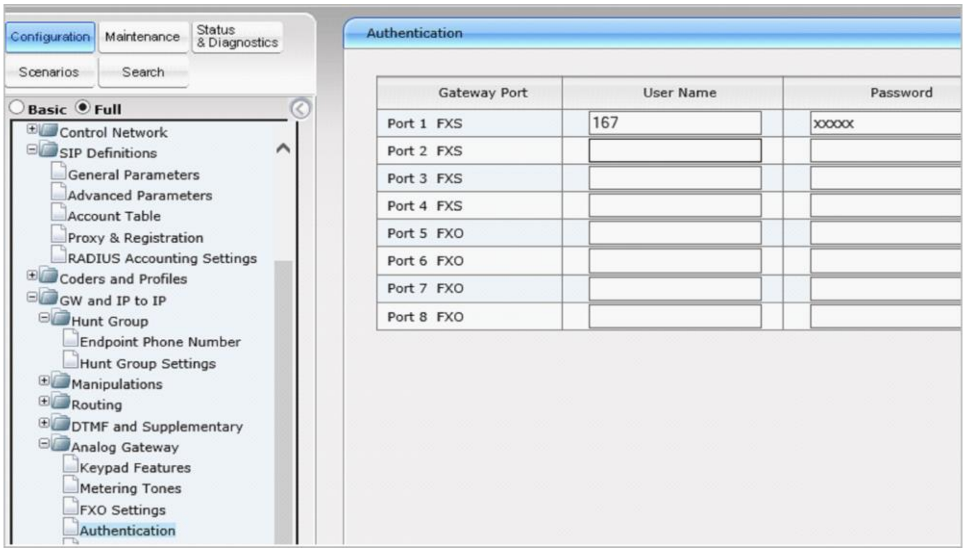


Figure 80: Re-entering the FXS extension authentication password if

1. (Optional) Save your current configuration as a precaution (skip this step if you don't need the old configuration). To save it, click **Maintenance** above the menu, and then select **Software Update** > **Configuration File**. Click **Save INI File** and choose a name and folder location.

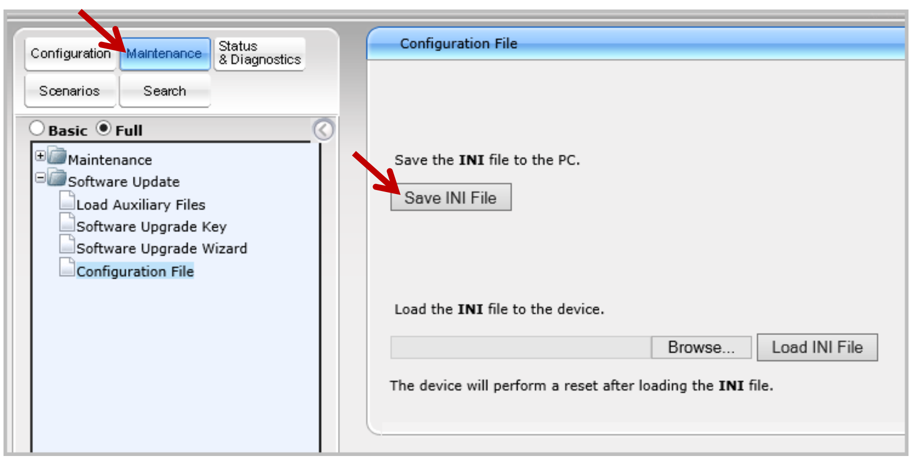


Figure 81: Save the current configuration

2. Reset the device to its default settings. This procedure will **not** reset the device's web login IP address.

   a. Use the Windows application Notepad to create an empty file. In this example, we will name it *null-config.ini*.

   b. In the AudioCodes configuration tool, click **Maintenance** above the menu, and then select **Software Update** > **Configuration File**. Click **Choose File** and select the empty file that you created in the previous step.

   c. Click **Load INI File** and follow the instructions to reboot the gateway.

Known Limitations and Workarounds

Following are known limitations and workarounds that you should consider while managing your gateway connections.

- Analog connections were validated and certified against firmware version 6.60A.265.010. Using other firmware versions could result in a loss of dial tone.
- If you power up an MP-118 gateway while it is not connected to the network, then you later attach the network connection or uplink switch, some of the channels may not be able to register.

  For this reason, you should make sure that all network cables and uplink switches are connected ***before*** you turn on your gateway.
- On an MP-118 gateway with a Level 3 uplink switch (such as Dell), it can take up to a full minute before the switch provides the network connection to the gateway. This delay is due to certain spanning tree settings on the switch, and may cause some channels not to register during the boot sequence.

  To avoid such problems, AltiGen recommends that you insert a regular switch between the Level 3 switch and the MP-118 gateway.
- If your network is not stable, the instability may cause some of the channels on the MP-124 to not register after a network outage.

AltiGen Technical Support

For FoIP implementation, AltiGen technical support will provide assistance and troubleshoot the configuration steps based on this configuration guide. Configurations other than the ones covered in this guide are not supported by AltiGen. If you encounter an issue with connectivity (for example, if the fax device does not drop the line), contact the device's manufacturer for support.

For general configuration information for your gateway device, 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