MAX Communication Server
Release 8.0

AudioCodes Mediant Gateway T1 Centrex Configuration Guide

October 5, 2015

Intended audience:

AltiGen Authorized Partners

Contents

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About This Guide

This guide describes how to configure T1 Centrex support for MAXCS 8.0 deployments using an AudioCodes Mediant series gateway.

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

- The examples in this guide illustrate the configuration for a Mediant 1000B gateway. For other models, adjust the port settings as needed. This document should also apply to Mediant 800B with a voice T1 interface.
- AltiGen does not provide general configuration support for AudioCodes devices. Contact AudioCodes support for assistance with general setup.
- T1 CAS Centrex transfer support is a special configuration for the Mediant 1000 gateway. You must dedicate the Mediant 1000 gateway to this purpose only. AltiGen will not support additional non-T1 CAS Centrex configuration to the same gateway. In other words, no additional PRI, non-Centrex T1 CAS and FXO, FXS ports should be used on this gateway. For additional FXS and FXO support, we recommend that you use an AudioCodes MP-1xx gateway.
- If non-Centrex PRI support is needed, check the AltiGen knowledgebase (https://know.altigen.com) for available PRI gateway options.

Requirements

Your system and environment must meet the following requirements:

- You must be using an AudioCodes gateway model Mediant 1000 with a T1/PRI module.
- You must have a support agreement with AudioCodes.
- The gateway must already have a static IP address and must be able to be configured through the web configuration tool. Make sure MaxCS can ping the gateway.
- The gateway must be running the correct version of firmware: Click the **Home** button and check that you have firmware version **F6.60A.292.001.** If the version is incorrect, talk to AudioCodes Support to obtain the correct firmware and update instructions.
- The gateway and MaxCS must be on the same LAN. If they are in different locations, a VPN must be set up between the two locations.

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
- AudioCodes gateway – 192.168.1.20

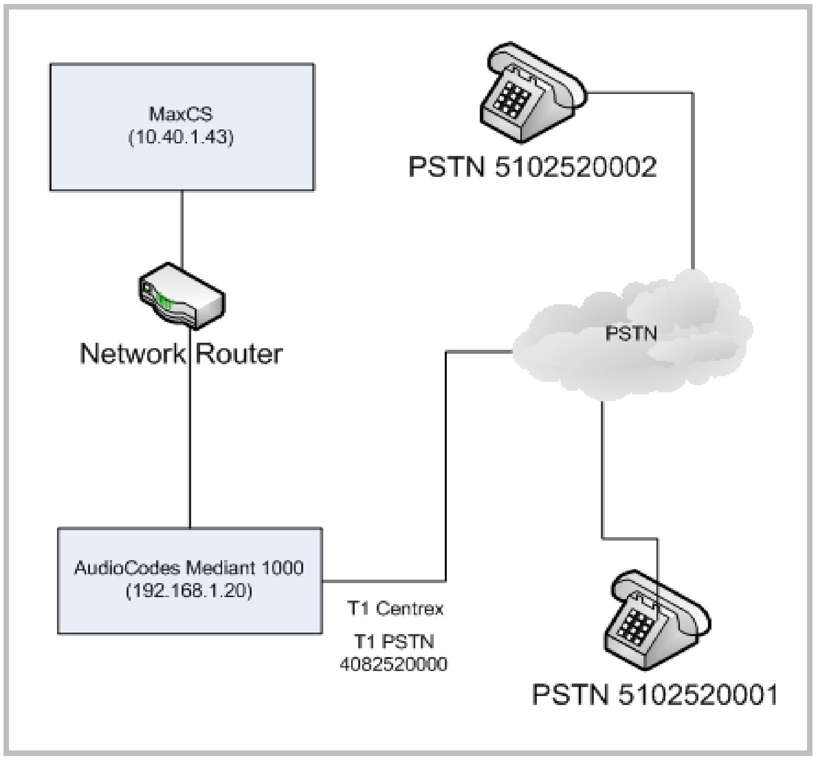


Figure 1: Example configurations for LAN deployments

In this example, T1 Centrex connects to CO (central office) that supports it.

It can also connect to a third party PBX that supports T1 Centrex. With a third-party PBX, these PSTN numbers can be replaced with extension numbers in the third-party PBX.

T1 Configuration

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

1. Your 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 27.
2. Next, obtain the appropriate CAS (channel associated signaling) file and INI file, to load to the gateway.

   If you are replacing an existing Max1000 or AltiGen T1 Centrex configuration with CO or a third-party PBX, then you may be able to use the CAS file that is attached to this article in the AltiGen knowledgebase. We recommend that you try to make the CO or third-party PBX T1 Centrex setting the same as specified in this document, so that you can directly use the CAS and INI file provided by AltiGen.

   Otherwise, you will need to contact AudioCodes to obtain the correct CAS file.

   AltiGen's T1 CAS file specifies the following ANI/DNIS formats:

   - AltiGen's T1 board uses DTMF to send ANI(caller ID) and DNIS(DID)

- The transmit and receive format is *ANI*DNIS

If you need to obtain a CAS file from AudioCodes, let them know that the file must support T1 CAS Centrex transfer with following parameters:

- LineTransferMode = 3
- TrunkTransferMode=3 [This value may change as a result of your discussions with AudioCodes Support]
- CASDelimitersPaddingUsage=1 [This value may change as a result of your discussions with AudioCodes Support]

The CAS file that is compatible with AltiGen T1 span is:

"e_m_fgbwinktable_hookflash(250)_ani_before_address.dat"

It is attached to this Article in the AltiGen knowledgebase.

If your CO third-party PBX Centrex T1 receives *ANI*DNIS format in DTMF, import the "BOARD_Manipulation.ini" file, so it can receive the call ID from gateway's T1 Centrex span.

If the format is different, talk to the CO/third-party PBX vendor to get the format. Then talk to AudioCodes support to get the correct .ini file.

3. Once you have obtained both the CAS and INI file, load them to the gateway.

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 after you have been idle for a few minutes; sometimes the application resets the menus to **Basic** mode.

a. Log into the AudioCodes configuration tool. Choose **Full** for the Menus.

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

b. Click the **Maintenance** button above the menu.

c. Select **Software Update** > **Load Auxiliary Files**.

d. Click **Browse** for the INI file, and load the appropriate tile. Repeat this step to load the CAS file.

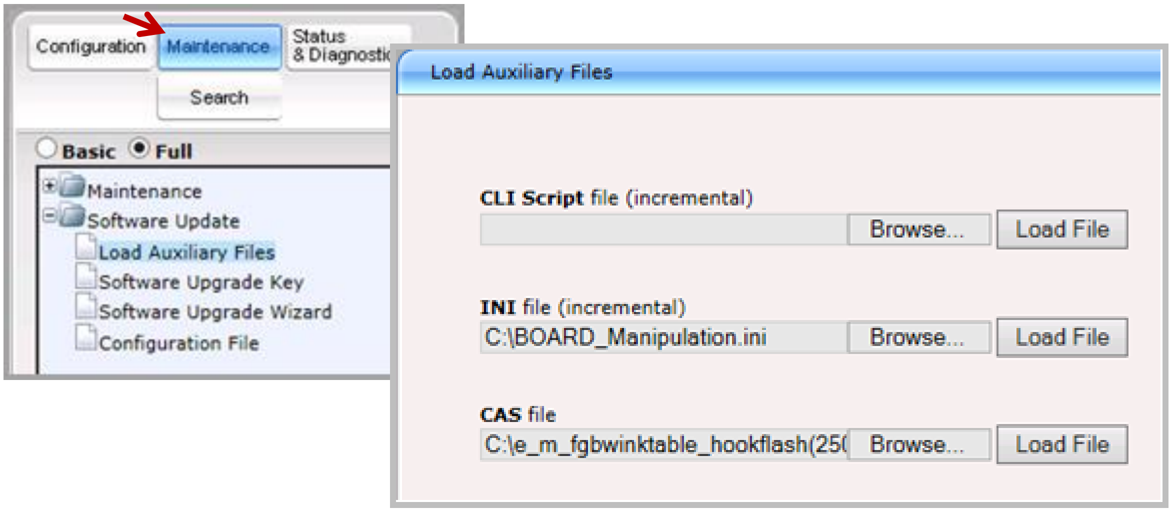


Figure 2: Load INI and CAS files to the gateway

Note: Do not load the same CAS file multiple times. If there are too many CAS files, you must remove the extra files. To do this, navigate to http://192.168.1.20/FAE, select **Cmd Shell**. Use either *DeleteCasFile 0* to delete the first CAS file, or use *DeleteCasFile -1* to delete all CAS files.

4. Select **VoIP** > **PSTN** > **CAS State Machines**.

   For AltiGen-compatible T1 Centrex systems, set the following parameters:

   - Set *Collect ANI* to **Yes**
   - Set *Digit Signaling System* to **DTMF**

   For other systems, set the parameters based upon the advice from AudioCodes.

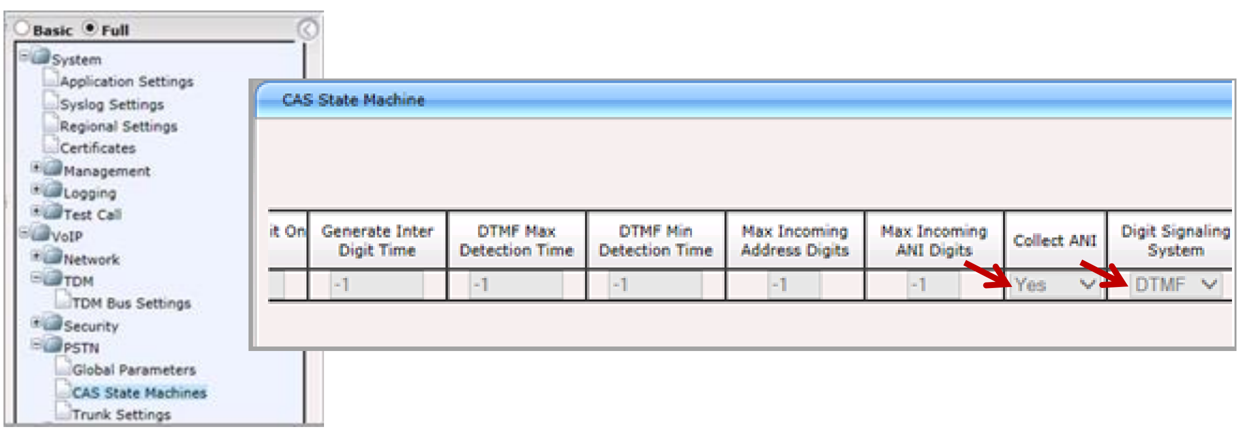


Figure 3: CAS State Machine parameters

5. Select **VoIP** > **PSTN** > **Trunk Settings**. Select the appropriate T1 span (Trunk ID). To do this, click on the number. For example, to select the second T1 span, click 2. In this document, we configure the first T1 span.

Figure 4: Click the trunk

6. The supported T1 interface parameters are as follows:

- Set *Protocol Type* **T1 CAS**
- Set *Line Code* to **B8ZS**
- Set *Framing Method* to **T1 FRAMING ESF CRC6**
- Set *CAS Table per Trunk* to the CAS file you just imported.

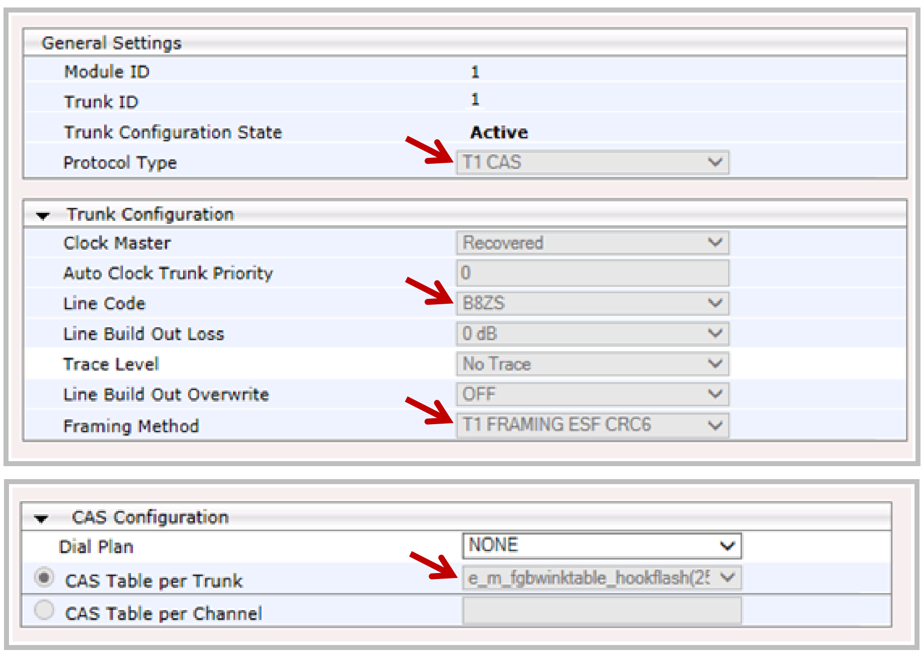


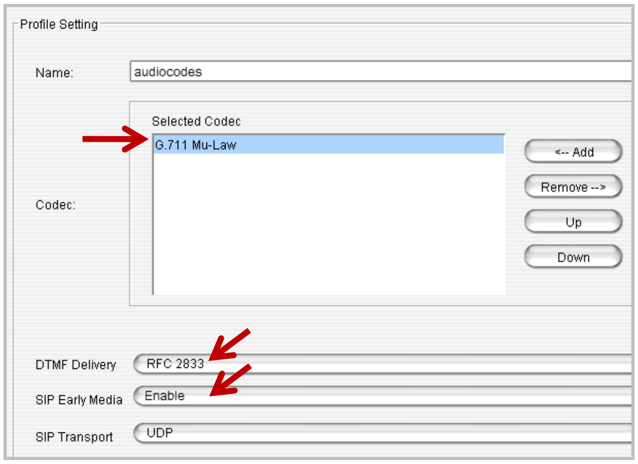
Figure 5: Set SIP Trunk parameters

7. Click **Apply Trunk Setting** to activate the span.

8. Check the CO or the third-party PBX status to confirm that the T1 span is up and running.

9. Log into MAXCS Administrator and select **VoIP** > **Enterprise Network Management** > **Codec** > **Codec Profile Table**.

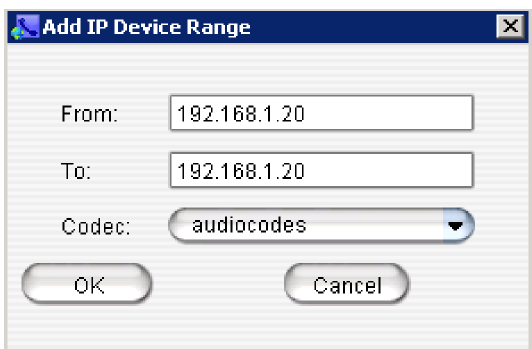
10. Add a new codec profile named *audiocodes*.



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

Figure 6: Add a new Codec Profile

11. Select **VoIP** > **Enterprise Network Management** > **Servers** > **IP Codec** > **IP Device Range**. Add the gateway's IP address.



- Set both the *From* and the *To* fields to 192.168.1.20
- For the *Codec*, select **audiocodes**

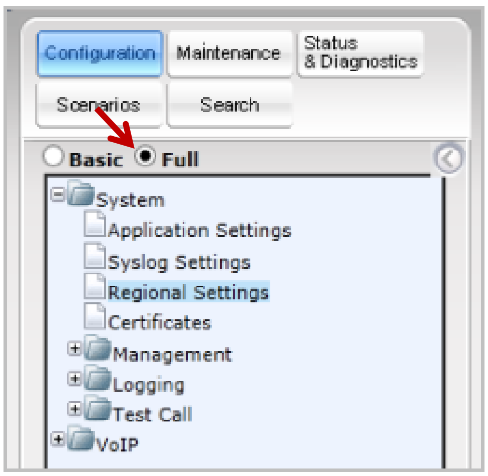
Figure 7: Add the Firewall IP address range

12. Open the MaxAdministrator *Boards* view. Double click **SIPSP**, select **Board Configuration**, and click **Advanced Configuration**. Add 192.168.1.20 to the *Trusted SIP Device List*.

Figure 8: Add the Firewall IP address to the Trusted Device list

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

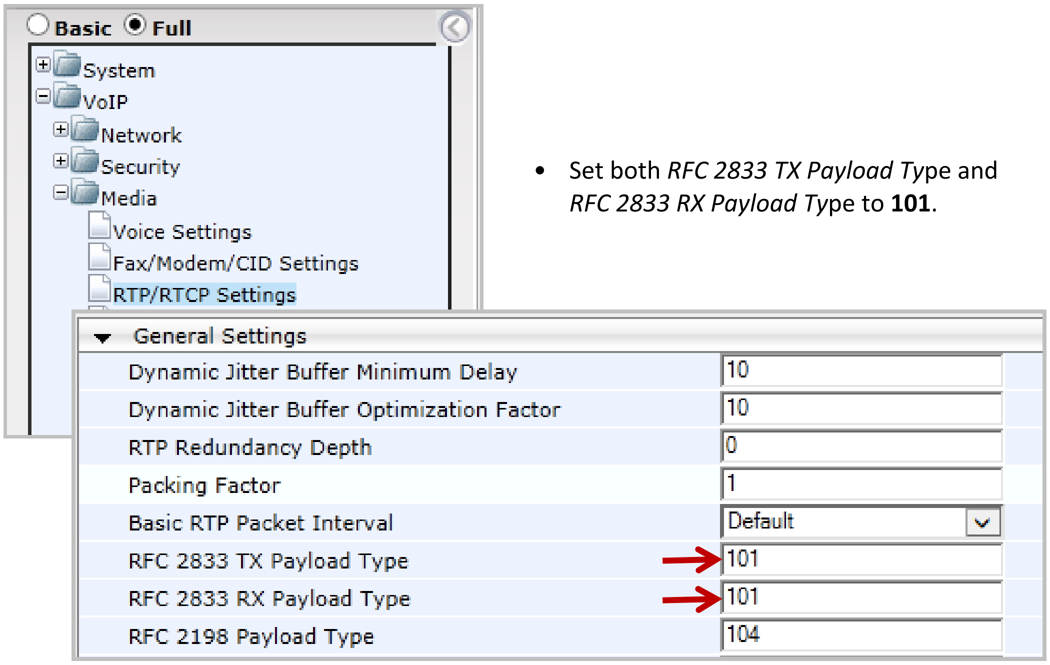
13. Open the AudioCodes configuration tool and log in.



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

Figure 9: Set the AudioCodes tool menu to Full

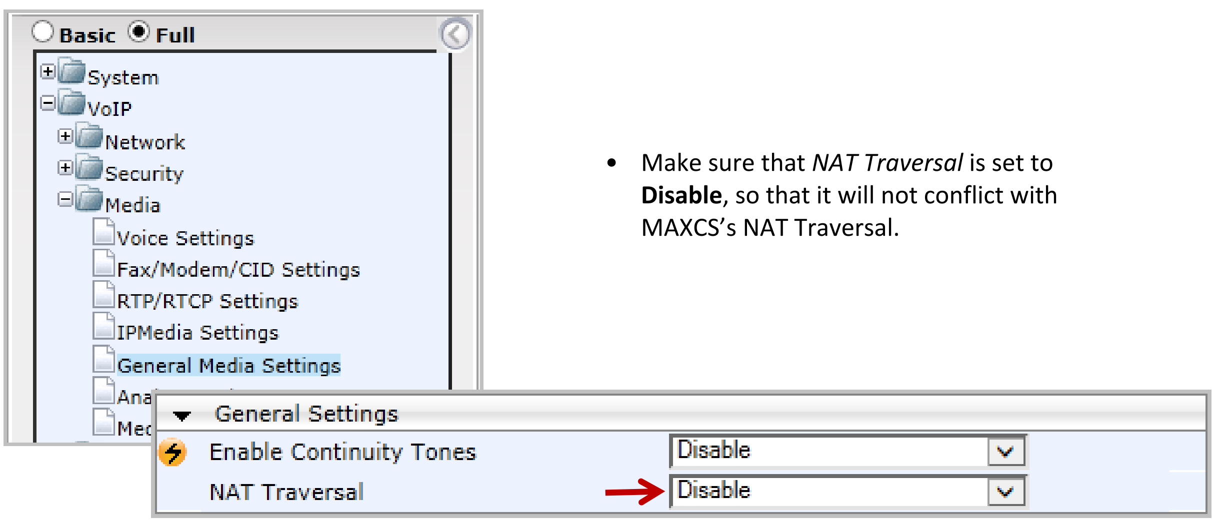
14. Select **VoIP** > **Media** > **RTP/RTCP Settings**.



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

Figure 10: Configure the TX and RX Payload Type settings

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

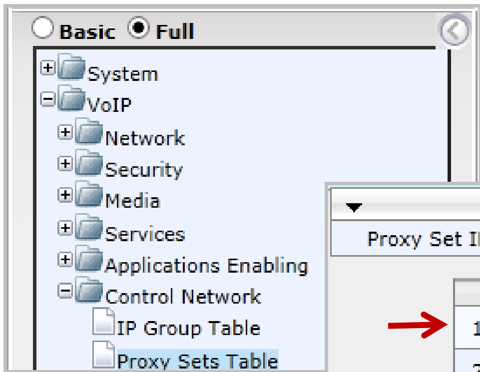


- Make sure that *NAT Traversal* is set to **Disable**, so that it will not conflict with MAXCS's NAT Traversal.

Figure 11: Disable NAT Traversal

16. 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**.



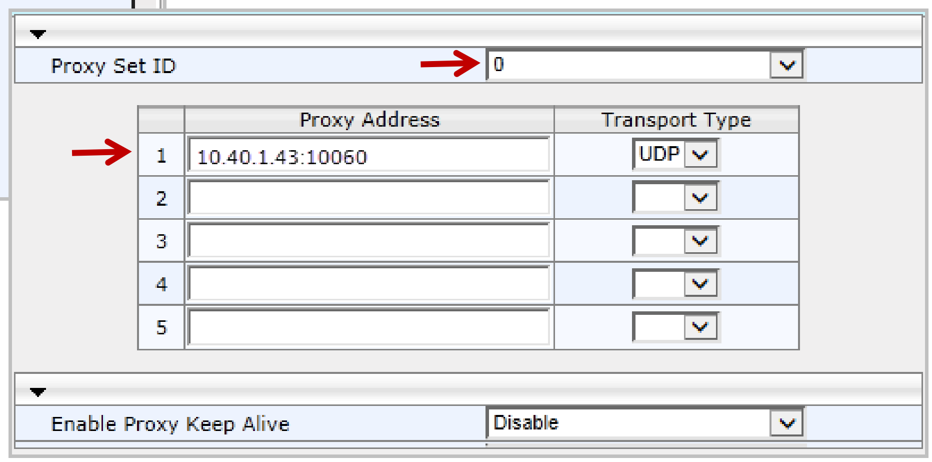


Figure 12: Add an entry to the Proxy Sets table

17. 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**.

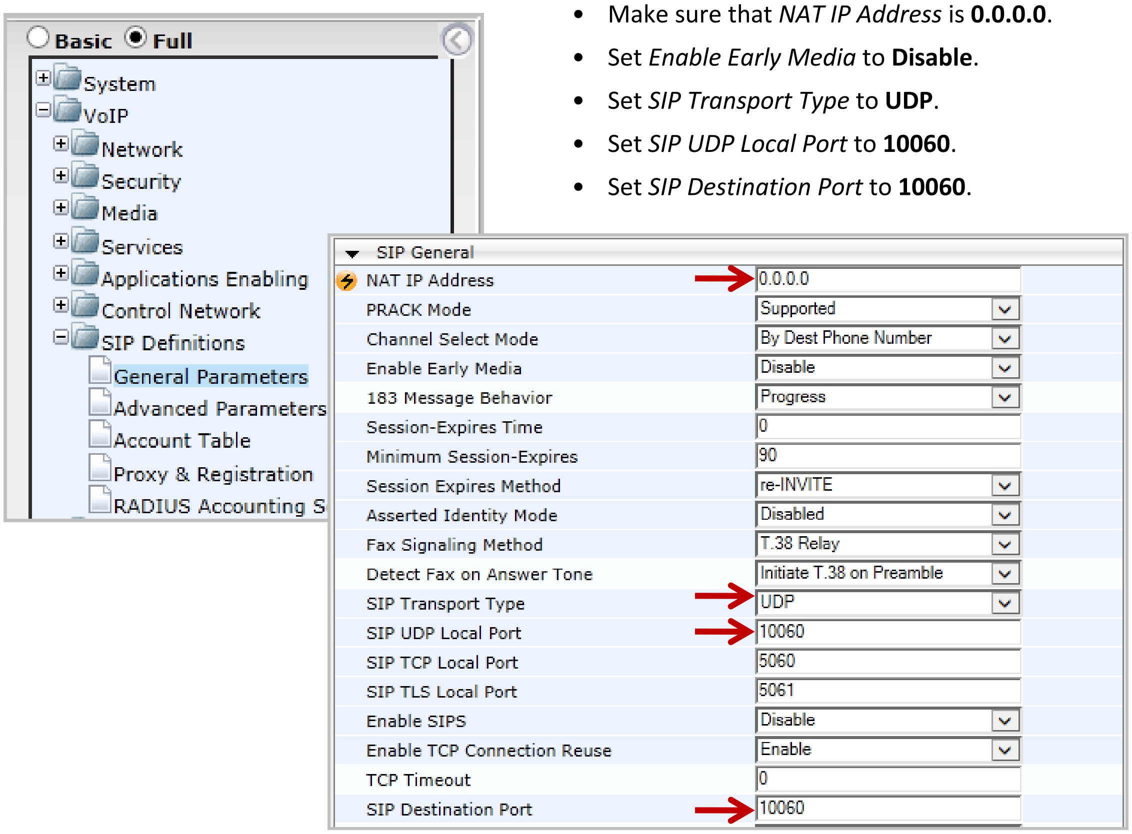
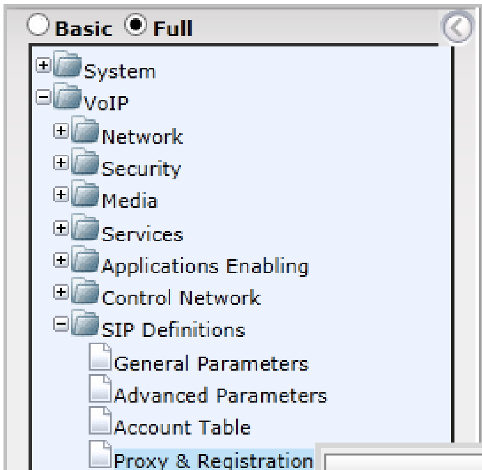


Figure 13: Configure general SIP parameters

18. 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**.
- Set *Registrar IP Address* to **10.40.1.43**. (The MAXCS IP address.)
- Set *Registrar Transport Type* to **UDP**.

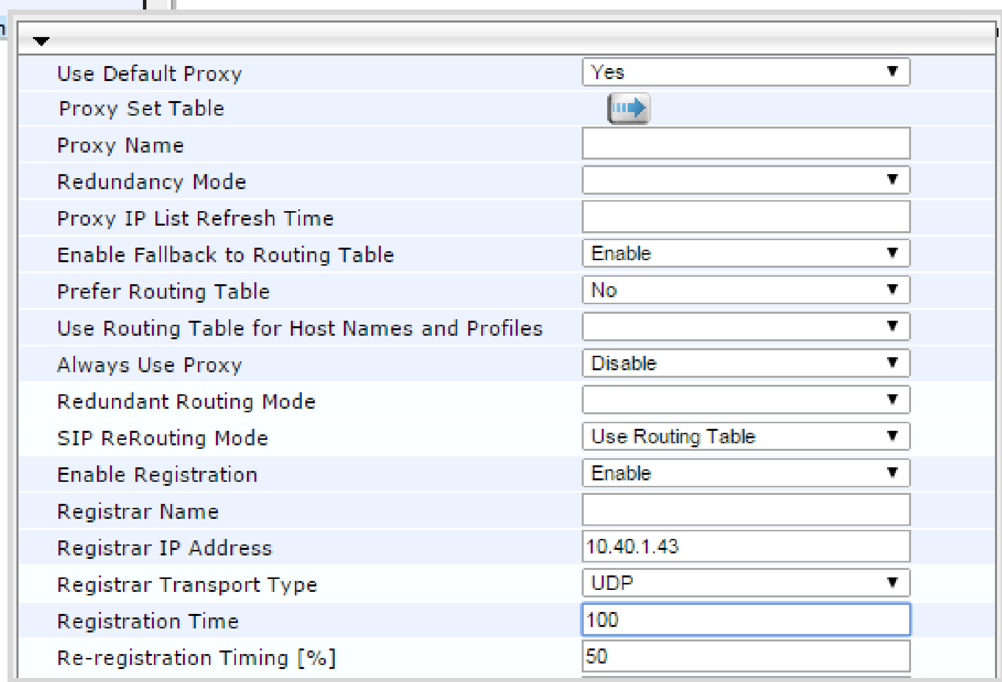


Figure 14: Configure Proxy and Registration parameters

19. 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.)

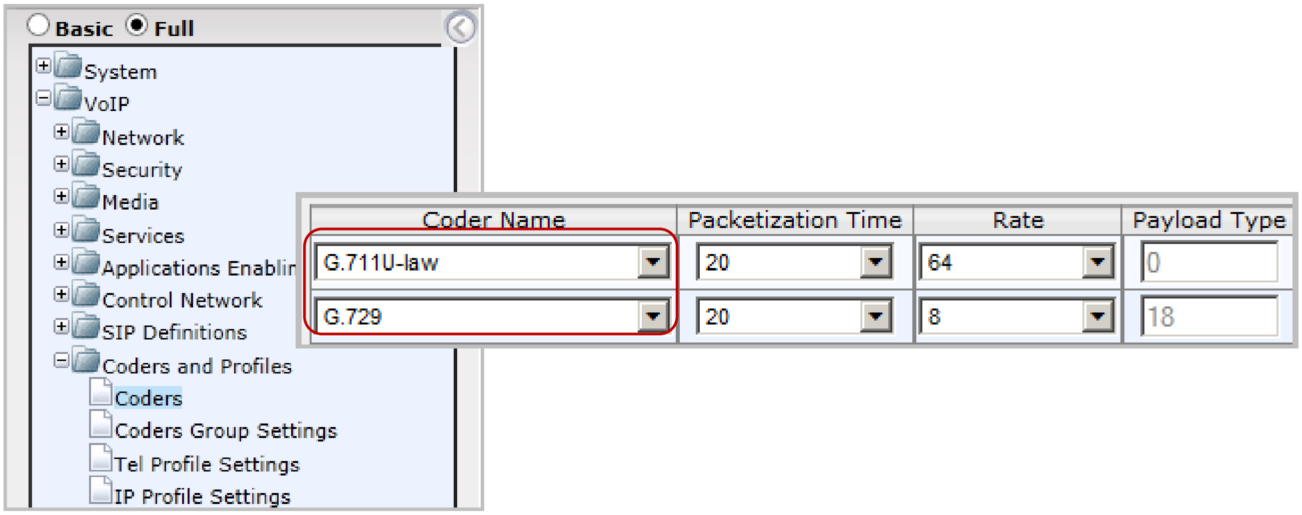


Figure 15: Confirm that G.711U-law and G.729 appear in the Coders list

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

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

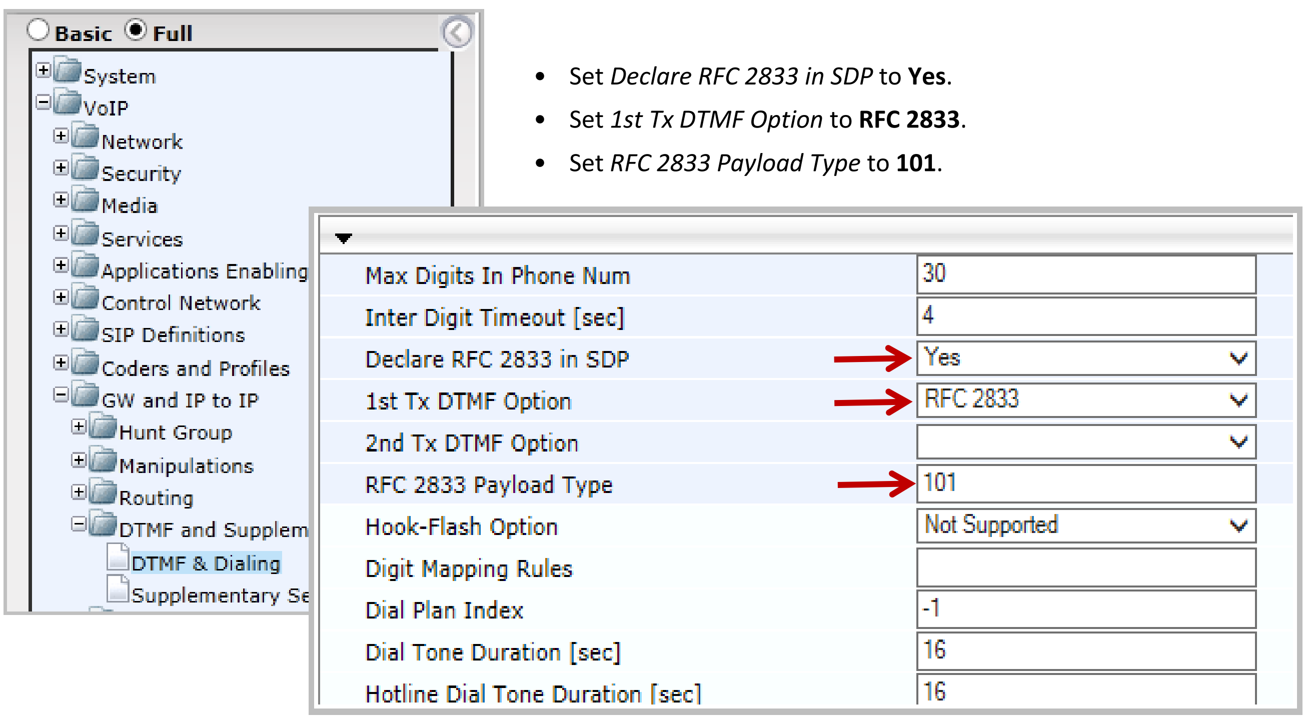
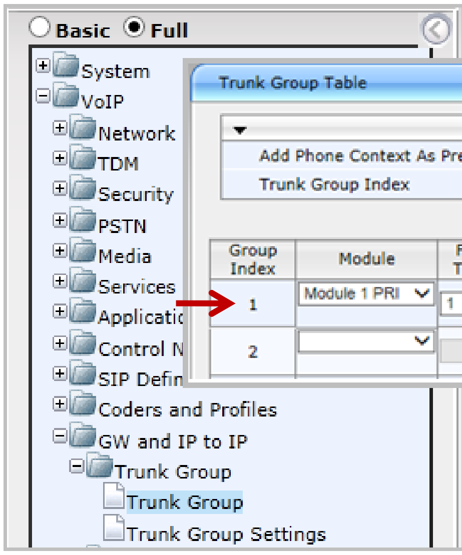
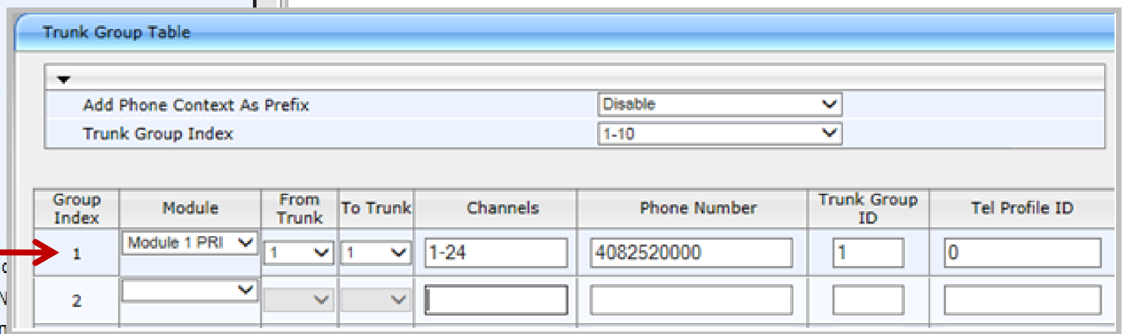


Figure 16: Configure DTMF and Dialing parameters

21. Select **VoIP** > **GW and IP to IP** > **Trunk Group** > **Trunk Group**. Configure Group Index 1 as follows:



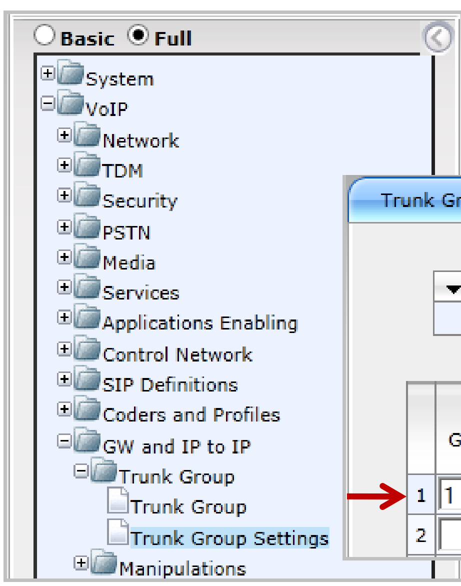


- Set *Module* to **Module 1 PRI**
- Set both *From Trunk* and *To Trunk* to **1**.
- Set *Channels* to **1-24**.
- Set *Phone Number* to your T1's PSTN number. In our example, this is 4082520000. If you do not have a number, enter any numbers.

Figure 17: Add trunk numbers to the Endpoint parameters

22. Select **VoIP** > **GW and IP to IP** > **Trunk Group** > **Trunk Group Settings**. Configure Group 1 as follows:

- Set *Trunk Group ID* to **1**.
- Set *Channel Select Mode* to **Cyclic Ascending**.
- Set *Registration Mode* to **Don't Register**.



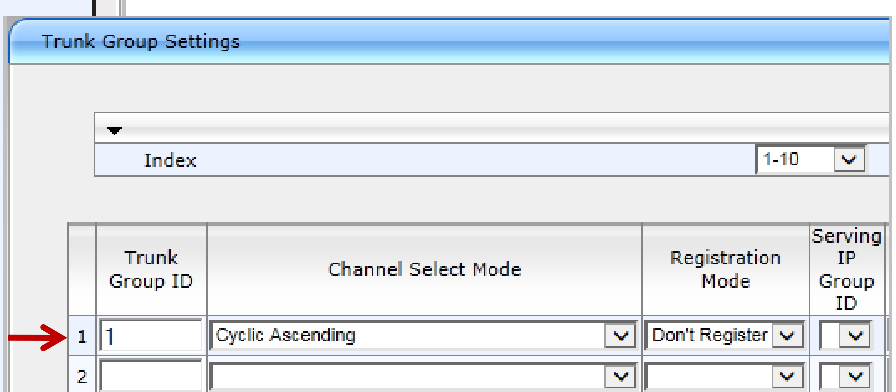


Figure 18: Configure Trunk Group 2

23. Select **VoIP** > **GW and IP to IP** > **Routing** > **IP to Trunk 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 address (in our example, this is 10.40.1.43), to prevent fraud calls. Do NOT enter an asterisk character in this field.
- Set *Trunk Group ID* to **1**

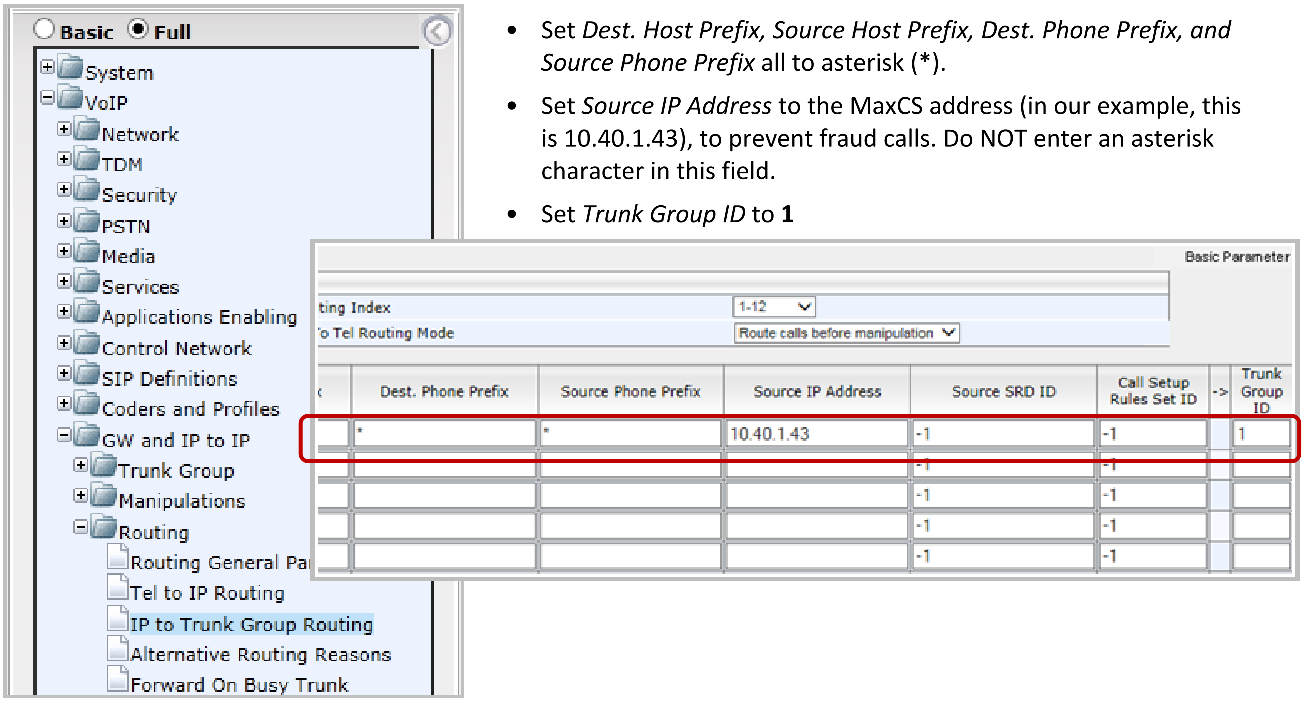


Figure 19: Configure a new IP to Trunk Group routing rule

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

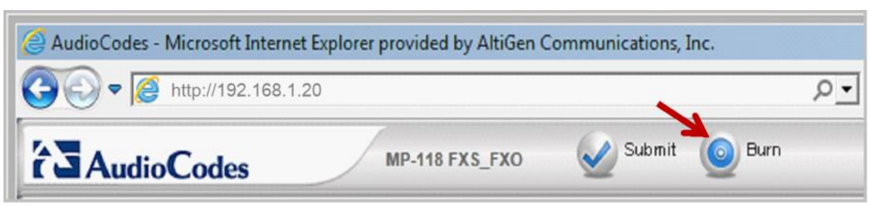


Figure 20: Click Burn to save the configuration changes

25. 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**.

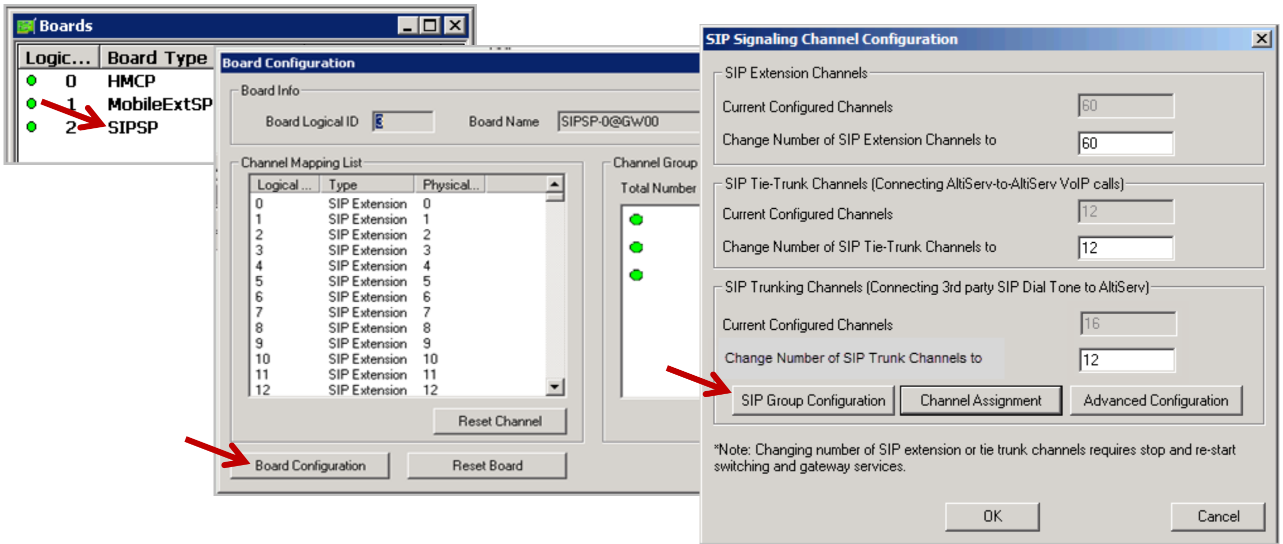


Figure 21: Open the SIP Group Configuration panel

26. First, create the SIP Group.

    Click **Add**. Enter a name for this SIP Group. Do not check the *AltiGen Trunk* option. Click **OK**.

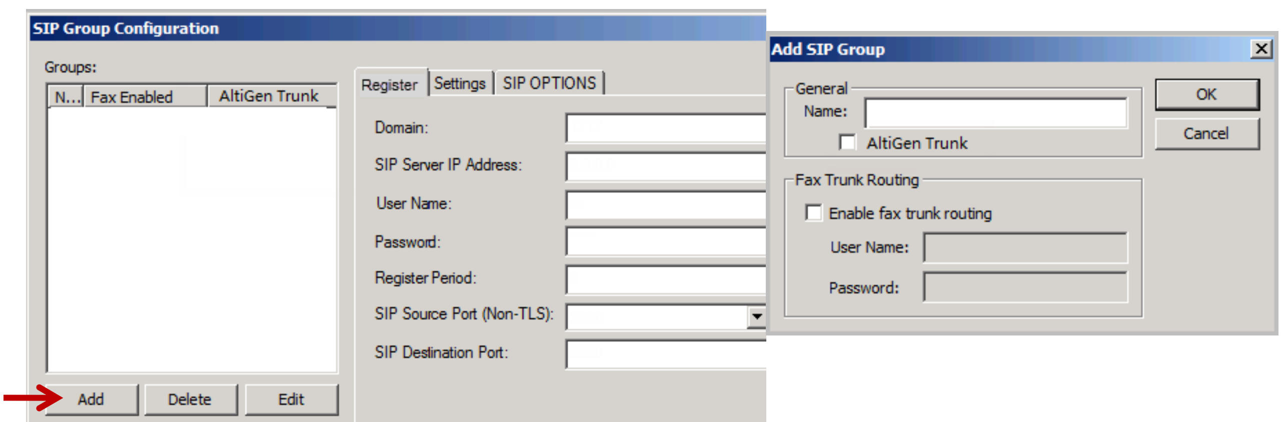


Figure 22: Add a new SIP Group

27. 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 *centrex*) and click the lower **Add** button (the one that is below the SIP Servers list). Enter the domain (this is usually the gateway's IP Address) and click **OK**.

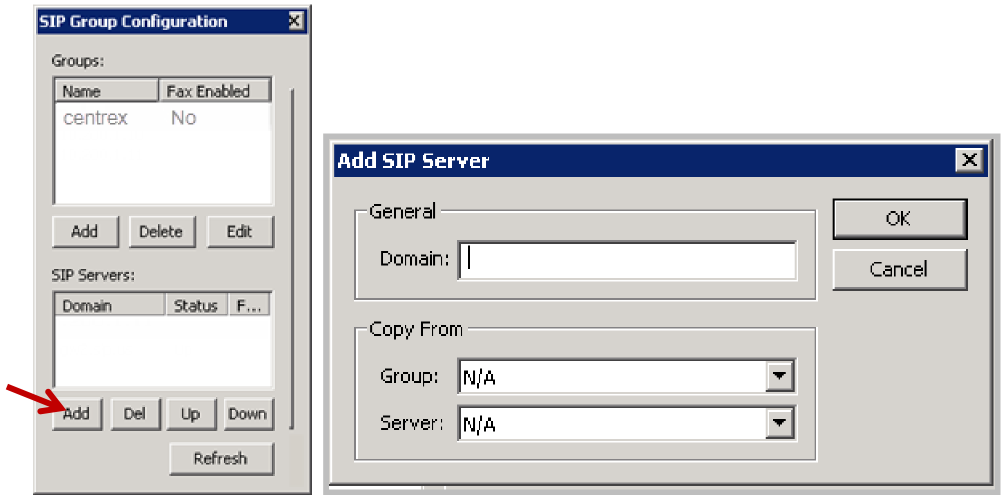


Figure 23: Add a SIP server to the new SIP Group

28. Highlight the new SIP Group. Select this new SIP Server in the lower window, and configure the settings on the Register tab as shown in the next figure. Enter the SIP Server IP address in the second field.

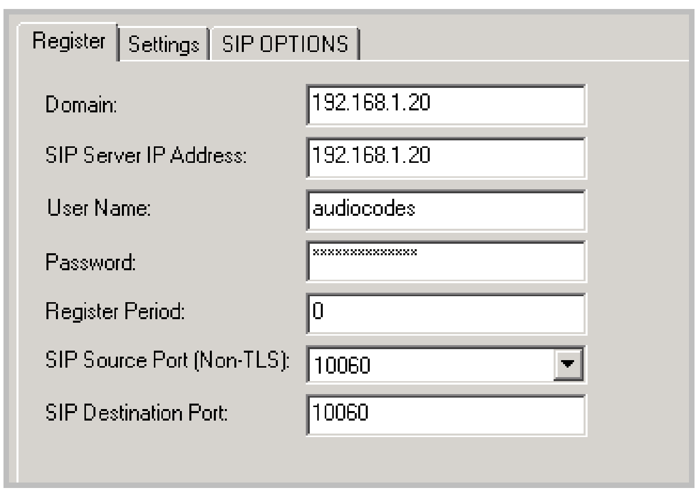


Figure 24: Configure the SIP server parameters

29. Next, enable channels. In 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**. Check their checkboxes to enable those channels.

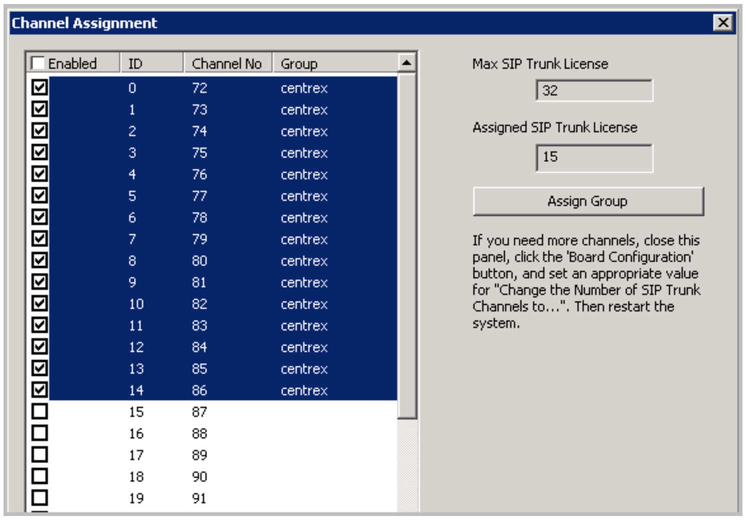


Figure 25: Enable channels

30. Add a trunk access code to the SIP Trunk that you just configured.
31. In your browser, navigate to this URL: http://192.168.1.20/AdminPage

    (Substitute the gateway's IP address for 192.168.1.20.)
32. On the left, select **ini Parameters**.
    - For the parameter *LINETRANSFERMODE*, enter **3**. Click **Apply New Value**.
    - For the parameter *TrunkTransferMode_0*, enter **3**. Click **Apply New Value**.

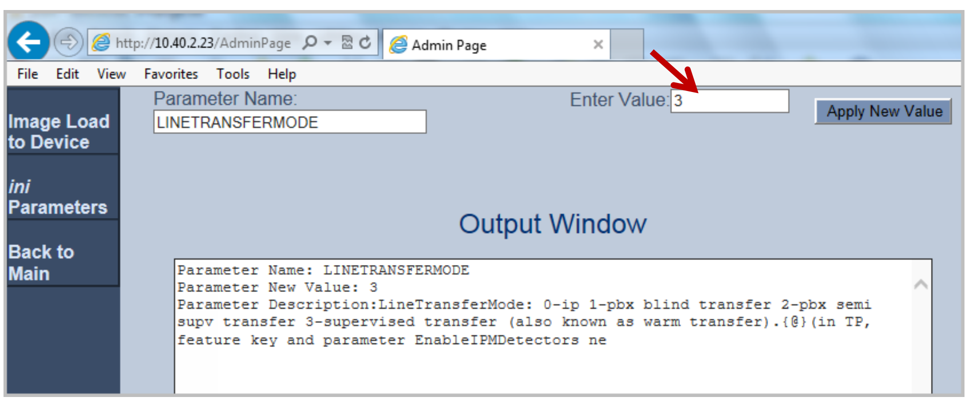


Figure 26: Specify INI parameters

Note: You can see that the *Trunk ID* shows as **1** in Figure 5**.**

TrunkTransferMode _0 means Trunk ID 1

TrunkTransferMode _1 means Trunk ID 2

TrunkTransferMode _2 means Trunk ID 3, and so on

33. For AltiGen-compatible T1 Centrex spans, for the parameter CASDelimitersPaddingUsage_0, enter **1**. Click **Apply New Value**. (*CASDelimitersPaddingUsage_0* means Trunk ID 1).

    For other systems, specify the value indicated by AudioCodes support.

34. Restart your gateway so that your changes can take effect.
35. At this point, test that you can make outbound calls and inbound calls from MAXCS using your IP phone and a T1 channel. Do not continue to the next section until these calls are working correctly.
36. Make sure the caller ID display is correct for inbound calls. If you are using a third- party PBX and you need the outbound caller id, confirm that your outbound caller ID display is also correct. Do not continue to the next section until these calls are working correctly.

T1 CAS Centrex Configuration

This configuration allows the system to release the T1 CAS channel after a transfer. For example, T1 channel 1 receives a call. The call is transferred. AudioCodes sends a Centrex flash to the same T1 channel to complete the transfer. Once the transfer is successful, the T1 channel is released and is ready to accept another call.

1. In the AudioCodes configuration tool, select VoIP > Media > IPMedia Settings. Set Answer Detector Activity Delay to 300.

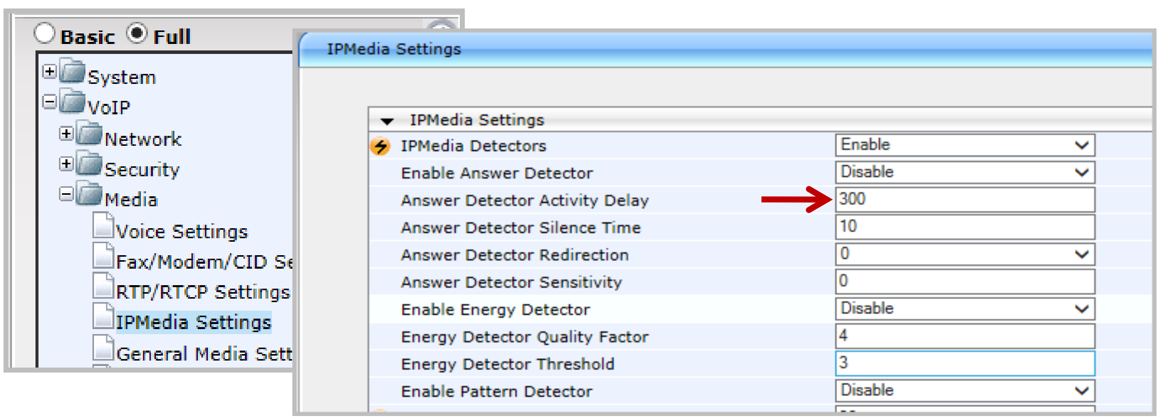
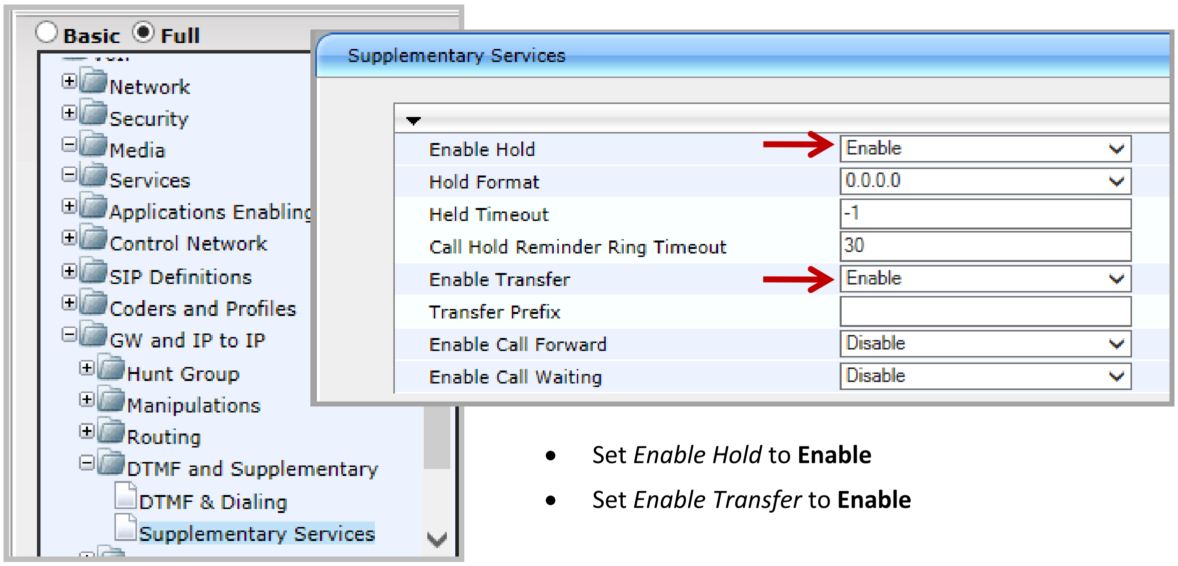


Figure 27: Adjust IP Media parameter

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



- Set *Enable Hold* to **Enable**
- Set *Enable Transfer* to **Enable**

Figure 28: Adjust Supplementary Services parameters

3. Log into MAXCS Administrator and double-click **SIPSP** in Boards view. Click **Board Configuration**. Click **SIP Group Configuration**.

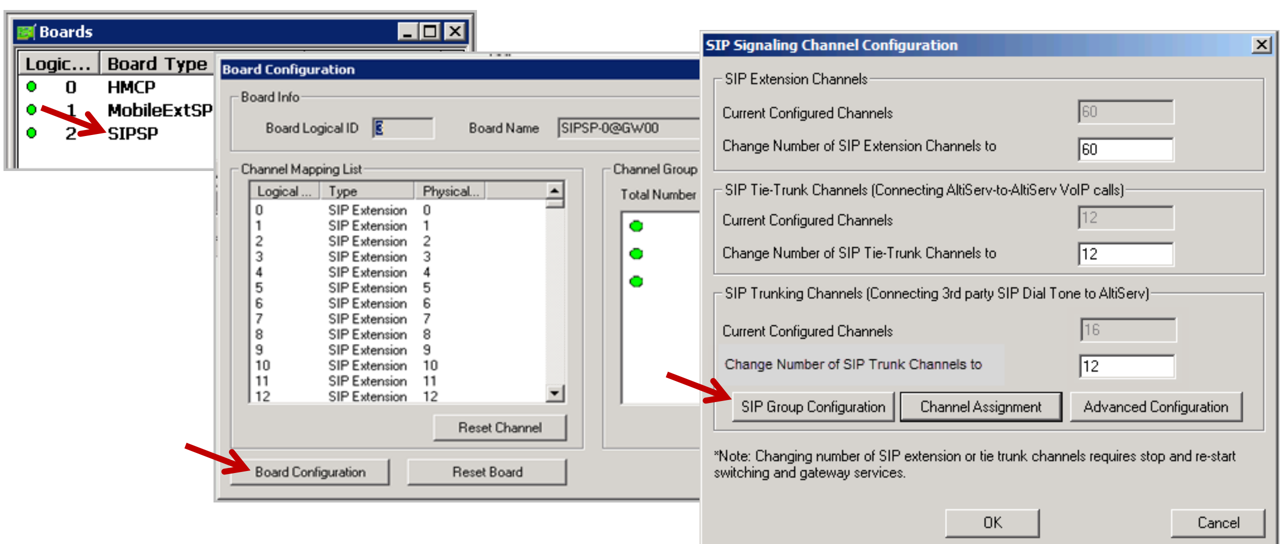


Figure 29: Open the SIP Group Configuration panel

4. First, create the SIP Group.

   Click **Add**. Enter a name for this SIP Group (you can use *centrex*, for example). Do not check the *AltiGen Trunk* option.

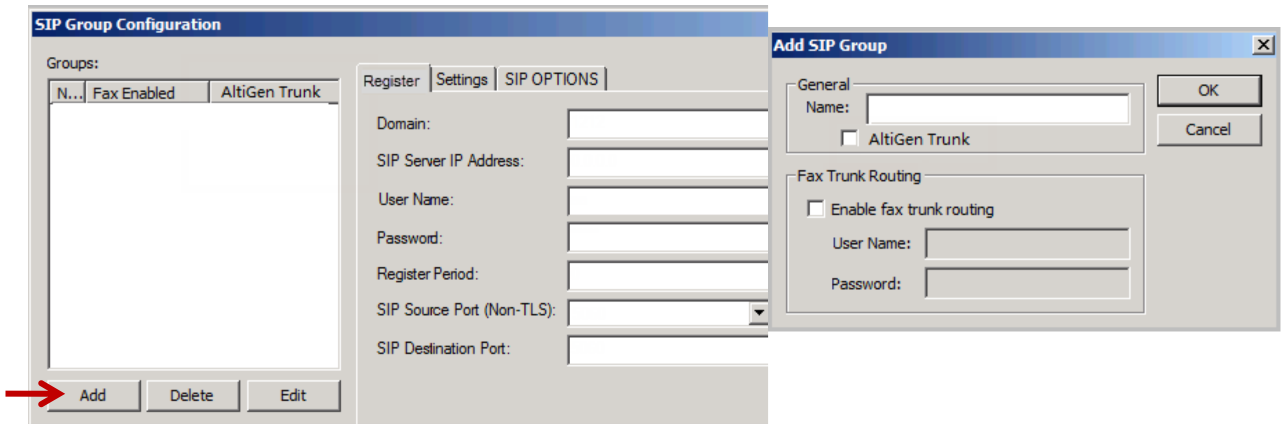


Figure 30: Add a new SIP Group

5. 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 name *centrex*) and click the lower **Add** button (the one that is below the SIP Servers list). Enter the URL for the domain and click **OK**.

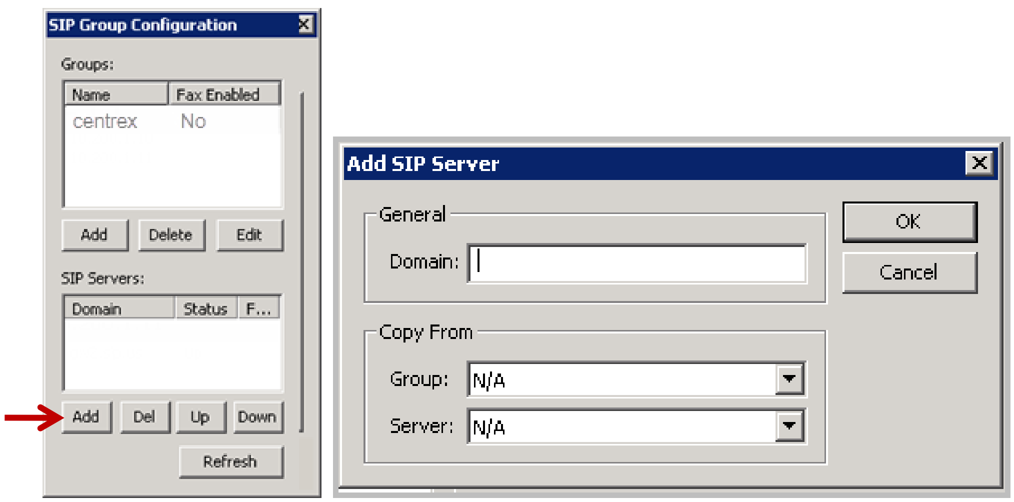


Figure 31: Add a SIP server to the new SIP Group

6. Highlight the new SIP Group. Select this new SIP Server in the lower window, and configure the settings on the *Register* tab as shown in the next figure. Enter the SIP Server IP address in the second field.

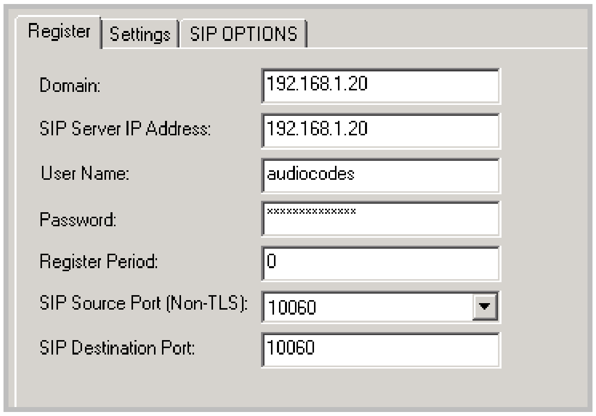


Figure 32: Configure the SIP server parameters

7. Switch to the *Settings* tab. Check both the *Enable SIP REFER* option and the *Enable Centrex Transfer* option.

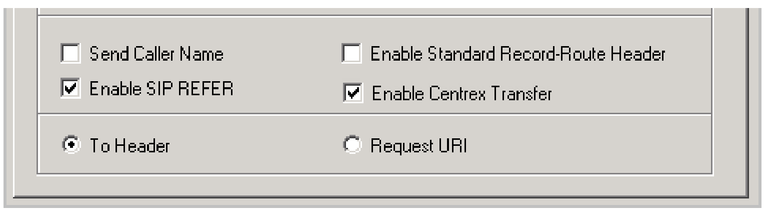


Figure 33: Enable two options on the Settings tab

8. Next, enable channels. Return to 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**. Check their checkboxes to enable those channels.

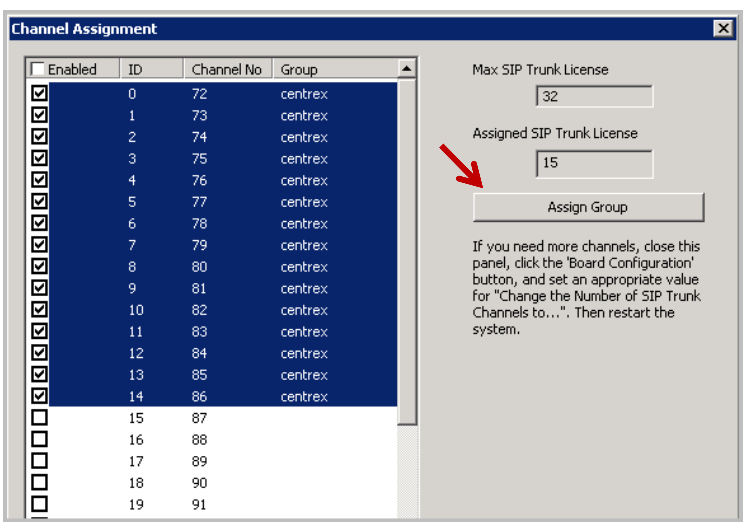


Figure 34: Enable channels

9. Add a trunk access code to the SIP Trunk that you just configured.

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

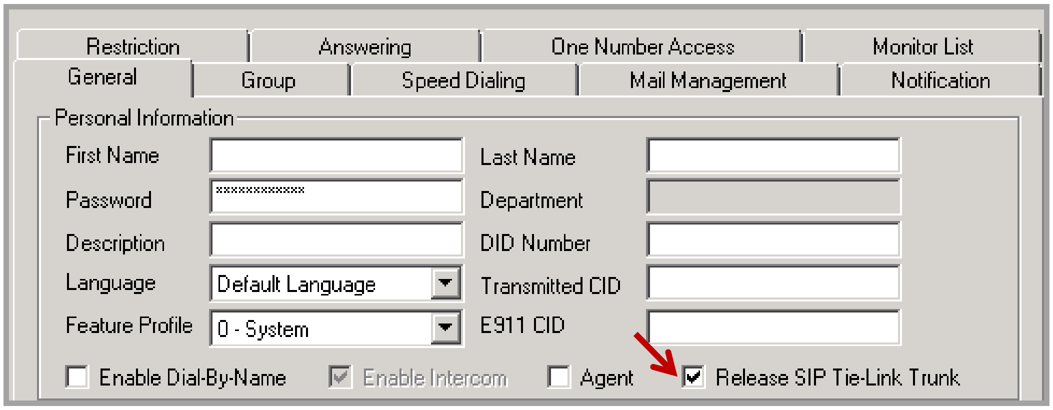


Figure 35: Configure extension 150

11. 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.

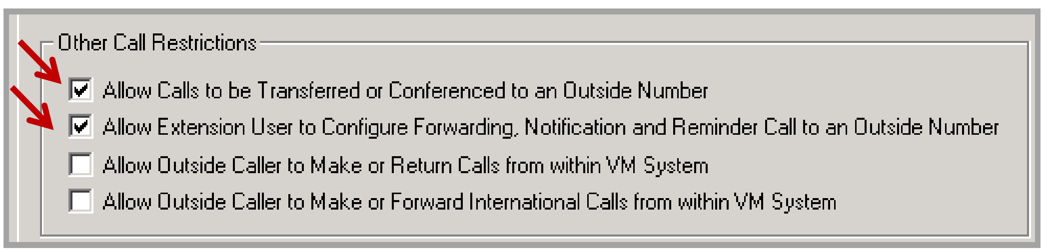


Figure 36: Set call restriction options for extension 150

12. Switch to the *Answering* tab. Check the *Enable Forward to* option 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.

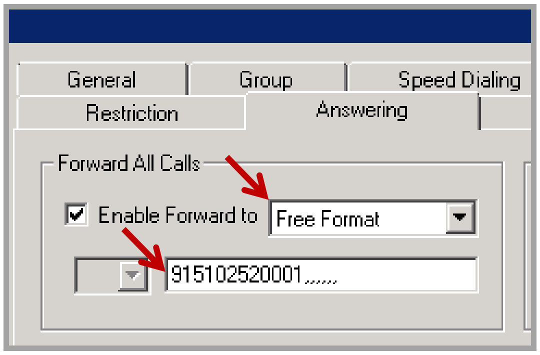


Figure 37: Set forwarding to Free Format

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

    a) From a mobile phone, make a call to the Mediant 1000 T1 channel. In this example, the number is 14082520000.

    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 released and ready to take the next call.

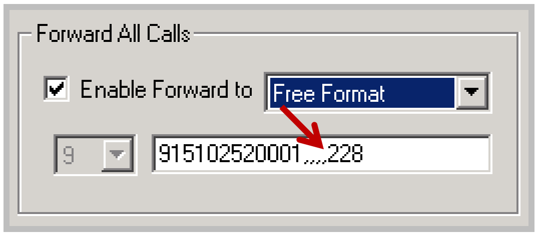


Figure 38: Inject numbers after the IVR system answers

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

Here is another example of when this feature can be useful. When an incoming call reaches an extension, the system will forward the call out and it will be released right away. Sometimes, the CO or PBX cannot respond in time, causing calls to drop. If this occurs, you can insert several prefix commas to give the CO or PBX more time to respond to Centrex transfers.

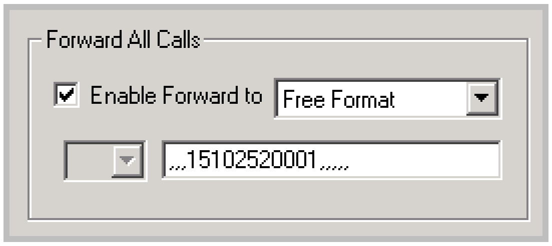


Figure 39: Insert commas to allow time for transfers to complete

Troubleshooting Tips

- 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.
- If MAXCS cannot correctly receive the Caller ID, confirm that *CASDelimitersPaddingUsage_X* is set correctly; X is the Trunk ID. (Refer to the steps on page 21.)

Resetting the Gateway to Default Settings

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

1. Save your current configuration as a precaution (optional). Click **Maintenance** above the menu, and then select **Software Update** > **Configuration File**. Click **Save INI File** and choose a name and folder location.

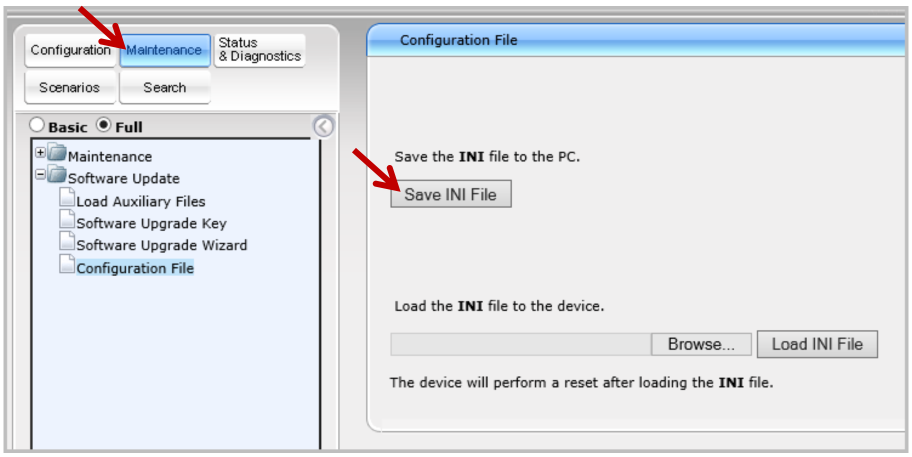


Figure 40: Save the current configuration

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 name it *nullconfig.ini*.
   b. In the AudioCodes configuration tool, click **Maintenance** above the menu and then select **Software Update** > **Configuration File**. Click **Browse** and select the empty file that you just created.
   c. Click **Load INI File** and follow the instructions to reboot the gateway.

About INI Files

Each model of Mediant gateway has a unique .INI file. For example, the .INI file for the Mediant 1000B is different from the .INI file for the model 800.

Because of this difference, you cannot take an .INI file created for a Mediant 1000B and use it as-is for a different model. You can, however, use an .INI file from one model and customize it for another model. To do this, copy the original .INI file and merge in the details for the other model.

AltiGen Technical Support

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.

For general configuration information for your gateway device, refer to your AudioCodes documentation. To find your gateway manual, contact AudioCodes to obtain a copy of the *Mediant 1000 SIP User's Manual*.

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