

MAX Communication Server Release 8.0

AudioCodes Mediant Gateway T1 Centrex Configuration Guide

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Intended audience: AltiGen Authorized Partners





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AltiGen Communications, Inc. 679 River Oaks Parkway San Jose, CA 95134 Telephone: 888-AltiGen (258-4436) Fax: 408-597-9020 E-mail: info@altigen.com Web site: www.altigen.com

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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 Audio-Codes MP-1xx gateway.
- If non-Centrex PRI support is needed, check the AltiGen knowledgebase (<u>https://know.altigen.com</u>) 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 <u>Resetting the Gateway to Default Settings</u> 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.

| Basic • Full Application Settings Syslog Settings Certificates Management Logging | CAS | State Machine | | | | | | |
|--|-------|------------------------------|----------------------------|----------------------------|--------------------------------|----------------------------|-------------|---------------------------|
| Test Call | it On | Generate Inter Digit Time | DTMF Max Detection Time | DTMF Min Detection Time | Max Incoming Address Digits | Max Incoming ANI Digits | Collect ANI | Digit Signaling System |
| BOTOM | | -1 | -1 | -1 | -1 | -1 | Yes 🗸 | DTMF 🗸 |
| CAS State Machines | | | | | | | | |

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.





| 1 2 | |
|---------|------|
| 0 19 20 | E DO |

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.

| Module ID | 1 | |
|----------------------------|---------------------------|---------|
| Trunk ID | 1 | |
| Trunk Configuration State | Active | |
| Protocol Type | T1 CAS | ~ |
| Trunk Configuration | | |
| Clock Master | Recovered | ~ |
| Auto Clock Trunk Priority | 0 | |
| Line Code | B8ZS | ~ |
| Line Build Out Loss | 0 dB | ~ |
| Trace Level | No Trace | \sim |
| Line Build Out Overwrite | OFF | ~ |
| Framing Method | T1 FRAMING ESF CRC6 | ~ |
| | | |
| CAS Configuration | | |
| Dial Plan | NONE | ~ |
| CAS Table per Trunk | e_m_fgbwinktable_hookflas | sh(2ť ∨ |
| and with the second second | | |

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.

| Profile Setting | | |
|-----------------|----------------|---------|
| Name: | audiocodes | |
| | Selected Codec | |
| _ | G.711 Mu-Law | < Add |
| | | Remove> |
| Codec: | | (Up) |
| | | Down |
| | | |
| | | |
| DTMF Deliver | y RFC 2833 | |
| SIP Early Med | ia Enable | |
| SIP Transport | UDP | |

- 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.

| 🚴 Add IP Devi | ce Range | × |
|---------------|--------------|---|
| | | |
| From: | 192.168.1.20 | |
| To: | 192.168.1.20 | |
| Codec: | audiocodes 🔍 | |
| ОК | Cancel | |

- 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 **Ad**-**vanced Configuration**. Add 192.168.1.20 to the *Trusted SIP Device List*.



| Adv | anced Configuration | |
|----------------|-------------------------|---|
| [¹ | Trusted SIP Device List | |
| | SIP Device IP Address | _ |
| | 10.140.0.22 | |
| | 10.140.0.23 | |
| | 10.140.0.24 | |
| | 10.140.0.27 | |
| | 10 140 0 30 | |
| | 192.168.1.20 | |
| | 192.168.1.152 | |

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.

| Configuration | Maintenance | Status & Diagnostics |
|---|--|-------------------------|
| Scenarios | Search | |
| O Basic 💿 F | Full | (|
| System Applica Syslog Region Certific Manaç Certific Manaç Test C U VoIP | ation Settings Settings al Settings cates gement ng Call | |

Figure 9: Set the AudioCodes tool menu to Full

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



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

| Basic ● Full System VoIP Wetwork © Security © Media Voice Settings Fax/Modem/CID Settings RTP/RTCP Settings | Set both RFC 2833 TX Payload Type and RFC 2833 RX Payload Type to 101. |
|---|--|
| | |
| Dynamic Jitter Buffer Minimum | Delay 10 |
| Dynamic Jitter Buffer Optimizat | tion Factor 10 |
| RTP Redundancy Depth | 0 |
| Packing Factor | 1 |
| Basic RTP Packet Interval | Default |
| RFC 2833 TX Payload Type | |
| RFC 2833 RX Payload Type | |
| RFC 2198 Payload Type | 104 |

Figure 10: Configure the TX and RX Payload Type settings

15. Select VolP > Media > General Media Settings.







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

| ● Basic ● Full ⊕ System ⊕ WoIP ⊕ Metwork ⊕ Security | | | Set Proxy Address 1 to MAXCS public IP addr Transport Type to UD | o 10.40.1 ress and p P . | . 43:10060 (The ort). Set its | |
|---|---------------|------|--|--|---|----------|
| ⊕ @ Media # @ Denviron | • | | | | | _ |
| Services Enabling | Proxy Set | t ID | \rightarrow | 0 | | ~ |
| Control Network | [| | Proxy Address | | Transport Type | |
| IP Group Table | \rightarrow | 1 | 10.40.1.43:10060 | | UDP 🗸 | |
| Proxy Sets Table | | 2 | | | ~ | |
| | | 3 | | | ~ | |
| | | 4 | | | ~ | |
| | | 5 | | | ~ | |
| | | | r | | | |
| | Enable Pr | oxy | Keep Alive | Disable | | ~ |

Figure 12: Add an entry to the Proxy Sets table



17. Select VolP > SIP Definitions > General Parameters.



Figure 13: Configure general SIP parameters



18. Select VolP > SIP Definitions > Proxy & Registration.

| System VoIP Network Security Media Services Control Network SIP Definitions General Parameters Advanced Parameters Account Table Proxy & Registration | Set Use Default Prox Set Enable Fallback to Set Prefer Routing To Set Always Use Prox Set SIP ReRouting M Set Enable Registrat Set Registrar IP Addit dress.) Set Registrar Transponent | ky to Yes . to Routing Table to Enable . Table to No . by to Disable . Tode to Use Routing Table . trion to Enable . tress to 10.40.1.43 . (The MAXCS IP ad- port Type to UDP . |
|---|--|--|
| | Use Default Proxy | Yes 🔻 |
| | Proxy Set Table | |
| | Proxy Name | |
| | Redundancy Mode | Υ |
| | Proxy IP List Refresh Time | |
| | Enable Fallback to Routing Table | Enable • |
| | Prefer Routing Table | No |
| | Use Routing Table for Host Names and Prof | files 🔹 🔻 |
| | Always Use Proxy | Disable T |
| | Redundant Routing Mode | ▼ |
| | SIP ReRouting Mode | Use Routing Table |
| | Enable Registration | Enable |
| | Registrar Name | |
| | Registrar IP Address | 10.40.1.43 |
| | Registrar Transport Type | UDP T |
| | Registration Time | 100 |
| | Re-registration Timing [%] | 50 |

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.)

| Basic Full System VoIP Network Security | | | | | |
|---|-------------|---|--------------------|------|--------------|
| Media | Coder Name | | Packetization Time | Rate | Payload Type |
| Generations Enablin | G.711U-law | - | 20 💌 | 64 💌 | 0 |
| | , [0.720 | | | | 10 |
| | G.729 | | 20 | lo 🚺 | 10 |
| Coders and Profiles | | | | | |
| Coders | | | | | |
| Coders Group Setti | ngs | | | | |
| Tel Profile Settings | | | | | |
| IP Profile Settings | | | | | |

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.



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:

| ⊖ Basic ● Full | | | 0 | | | | | |
|----------------------------------|----------------|--|--|--|-------------------------------------|---|---------------------------|-----------------------|
| System | Trunk Gr | oup Table | | | | | | |
| • Metwork | - | | | | | | | |
| ±∭трм | Add | Phone Context As | Prefix | | | Disable | ~ | |
| ∎ | Trur | nk Group Index | | | | 1-10 | ~ | |
| €@PSTN | | | | | | | | |
| ⊕@ Media | Group Index | Module | From Trunk | To Trunk | Channels | Phone Number | Trunk Group ID | Tel Profile ID |
| Services Deplication | 1 | Module 1 PRI V | 1 🗸 | 1 🗸 | 1-24 | 4082520000 | 1 | 0 |
| E Control N | 2 | × | ~ | ~ | | | | |
| € Coders and | l Profiles | | | | | | | |
| GW and IP to IP GMTrunk Group | | | Set <i>Module</i> to Module 1 PRI | | | | | |
| | | • Set both <i>From Trunk</i> and <i>To Trunk</i> to 1 . | | | | | | |
| Trunk Group Settings | | | | • Set <i>Channels</i> to 1-24 . | | | | |
| | | | | Set ple | <i>Phone Numbe</i> , this is 408252 | r to your T1's PSTN ni 0000. If you do not h | umber. In o ave a numb | ur exam- er, enter |

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.

any numbers.

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:

| ● Basic ● Full | © | • | Set Dest. Host Pr Source Phone Pre | <i>efix, Source Host efix</i> all to asterisk | Prefix, Dest. Pho (*). | one Prefix, a | nd | |
|--|-----------------------------------|--------|---|--|--|------------------------------|-----------|----------------------|
| VoIP TDM Control C | | • | Set Source IP Add is 10.40.1.43), to character in this Set Trunk Group | dress to the Max prevent fraud ca field. ID to 1 | CS address (in ou alls. Do NOT ente | ir example, er an asteris | this k | |
| ■ Media ■ Services ■ Applications Enabling ■ Control Network | ting Index 'o Tel Routing Mode | | | 1-12 Route calls before maniput | lation 🗸 | Ba | sic Para | amete |
| | C Dest. Phone | Prefix | Source Phone Prefix | Source IP Address | Source SRD ID | Call Setup Rules Set ID | -> 0 | frunk Group ID |
| GW and IP to IP | * | | . | 10.40.1.43 | -1 | -1 |] [1 | |
| 🗉 🗐 Trunk Group | | | Ŭ | | - 1 | -1 | | _ |
| • Manipulations | | | <u> </u> | | -1 | -1 | | |
| □ 🗇 Routing | | | | | -1 | -1 | | |
| Routing General Pa | | | ļ | | -1 | -1 | | |
| Tel to IP Routing IP to Trunk Group R Alternative Routing Forward On Busy Tr | outing Reasons runk | | | | | | | |

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.

| 🧟 AudioCodes - Microsoft Intern | et Explorer provided by AltiGen C | Communications, Inc. | |
|---------------------------------|-----------------------------------|----------------------|------|
| COC - C http://192.168.1 | .20 | | • 0 |
| | MP-118 FXS_FXO | Submit 🧕 i | Burn |

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**.



| 🛒 Boards | | [| STP Signaling Channel Configuration | X |
|--|---|---|--|--------|
| Logic Board Type • 0 HMCP • 1 MobileExtSP • 2 SIPSP | Board Configuration Board Logical ID Board Name SIF Channel Mapping List Cogical Type Physical 0 SIP Extension 0 1 1 2 SIP Extension 1 2 3 SIP Extension 2 3 3 SIP Extension 3 4 SIP Extension 3 4 5 SIP Extension 5 6 SIP Extension 7 8 SIP Extension 7 8 SIP Extension 10 11 SIP Extension 11 12 SIP Extension 11 12 SIP Extension 11 12 SIP Extension 12 Tester Channel 8 8 8 12 SIP Extension 12 Tester Channel 8 8 8 12 Tester Channel 12 12 12 12 12 12 12 12 12 12 12 14 14 14 14 14 14 14 14 14 14 14 14 14 14 14 | PSP-0@GW00 Channel Group Total Number • • | SIP Signaling Channel Configuration SIP Extension Channels Current Configured Channels Change Number of SIP Extension Channels to SIP Tie-Trunk Channels (Connecting Alt/Serv-to-Alt/Serv VolP calls) Current Configured Channels Current Configured Channels Current Configured Channels Current Configured Channels SIP Trunking Channels (Connecting 3rd party SIP Dial Tone to Alt/Serv) Current Configured Channels Current Configured Channels Current Configured Channels Current Configured Channels SIP Trunking Channels (Connecting 3rd party SIP Dial Tone to Alt/Serv) Current Configured Channels Change Number of SIP Trunk Channels to SIP Group Configuration Channel Assignment Advanced Configure "Note: Changing number of SIP extension or tie trunk channels requires stop and reswitching and gateway services. | ation |
| La La | | | ΟΚ | Cancel |

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.

| SIP Group Configuration | | | |
|-----------------------------|-------------------------------|--------------------------|--------|
| Groups: | | Add SIP Group | × |
| N Fax Enabled AltiGen Trunk | Register Settings SIP OPTIONS | General | ОК |
| | Domain: | Name: | Cancel |
| | SIP Server IP Address: | Fax Trunk Routing | |
| | User Name: | Enable fax trunk routing | |
| | Password: | User Name: | |
| | Register Period: | Password: | |
| | SIP Source Port (Non-TLS): | | |
| | SIP Destination Port: | | |
| Add Delete Edit | | | |

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**.



| Name Fax Enabled | | |
|------------------|-------------------------|--------------|
| | Add SIP Server | |
| Add Delete Edit | General Domain: | OK Cancel |
| Domain Status F | Copy From Group: N/A | |
| Add Del Up Down | Server: N/A | |

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.

| Register Settings SIP OPTIONS | | | | |
|-------------------------------|--------------|--|--|--|
| Domain: | 192.168.1.20 | | | |
| SIP Server IP Address: | 192.168.1.20 | | | |
| User Name: | audiocodes | | | |
| Password: | ******* | | | |
| Register Period: | 0 | | | |
| SIP Source Port (Non-TLS): | 10060 | | | |
| SIP Destination Port: | 10060 | | | |
| | | | | |

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.
- In your browser, navigate to this URL: <u>http://192.168.1.20/AdminPage</u> (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.

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

| IPMedia Settings | | |
|--------------------------------|------------------|--|
| ✓ IPMedia Settings | | |
| IPMedia Detectors | Enable | ~ |
| Enable Answer Detector | Disable | ~ |
| Answer Detector Activity Delay | 300 | |
| Answer Detector Silence Time | 10 | |
| Answer Detector Redirection | 0 | ~ |
| Answer Detector Sensitivity | 0 | |
| Enable Energy Detector | Disable | ~ |
| Energy Detector Quality Factor | 4 | |
| Energy Detector Threshold | 3 | |
| Enable Pattern Detector | Disable | ~ |
| | IPMedia Settings | IPMedia Settings ✓ IPMedia Detectors Enable Answer Detector Disable Answer Detector Activity Delay 300 Answer Detector Silence Time 10 Answer Detector Redirection 0 Answer Detector Sensitivity 0 Enable Energy Detector Disable Energy Detector Quality Factor 4 Energy Detector Threshold 3 Enable Pattern Detector Disable |

Figure 27: Adjust IP Media parameter

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





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.

| 🛒 Boards 📃 🗆 🗙 | STP Signaling Channel Configuration |
|---|--|
| Logic Board Type Board Configuration 0 HMCP 1 MobileExtSP 0 2 SIPSP Board Logical ID Board Name SIPSP-0@GW00 Channel Mapping List Channel Mapping List Channel SiPSP-0@GW00 Total Number 0 SIP Extension 1 SIP Extension 1 1 SIP Extension 1 1 SIP Extension 1 2 SIP Extension 1 3 SIP Extension 3 4 3 SIP Extension 3 4 SIP Extension 6 6 7 SIP Extension 6 7 SIP Extension 7 8 SIP Extension 8 9 SIP Extension 10 11 SIP Extension 10 11 SIP Extension 10 11 SIP Extension 11 12 File Extension 12 Image: SIP Extension 10 11 SIP Extension 11 12 Image: SIP Extension 12 Image: SIP Extension 12 Image: SIP Extension 11 12 Image: SIP Extension 12 Image: SIP Extension < | SIP Extension Channels 60 Current Configured Channels 60 Change Number of SIP Extension Channels to 60 SIP Tie-Trunk Channels (Connecting AltiServ-to-AltiServ VolP calls) |

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.



| SIP Group Configuration | | | | | |
|---|---|-----|---|----|--------------|
| SIP Group Configuration Groups: N Fax Enabled AltiGen Trunk | Register Settings SIP OPTI Domain: SIP Server IP Address: | ons | Add SIP Group General Name: AltiGen Trunk Fax Trunk Routing | | ОК Сапсеl |
| | User Name: Password: Register Period: SIP Source Port (Non-TLS): | | Enable fax trunk routi User Name: Password: | ng | |
| Add Delete Edit | SIP Destination Port: | | | | |

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**.

| Name | Fax Enabled | | |
|-------------|-------------|----------------|--------|
| centrex | No | | |
| | | Add SIP Server | |
| Add D | | General | ок |
| IP Servers: | | Domain: | Cancel |
| Domain | Status F | | |
| | | Copy From | |
| | | Group: N/A | |
| Add Del | Up Down | Server: N/A | |
| | Refresh | | |

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.

| Register Settings SIP OPTIONS | | | | |
|-------------------------------|--------------|--|--|--|
| Domain: | 192.168.1.20 | | | |
| SIP Server IP Address: | 192.168.1.20 | | | |
| User Name: | audiocodes | | | |
| Password: | ******** | | | |
| Register Period: | 0 | | | |
| SIP Source Port (Non-TLS): | 10060 💌 | | | |
| SIP Destination Port: | 10060 | | | |
| | | | | |

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.

| Send Caller Name Enable SIP REFER | Enable Standard Record-Route Header Enable Centrex Transfer | |
|--|---|--|
| To Header | C Request URI | |

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.

| Restriction | Answering | One Number Access Monitor List | | |
|--|------------------|---|--|--|
| General | Group Spee | ed Dialing Mail Management Notification | | |
| Personal Information | | | | |
| First Name | | Last Name | | |
| Password | ***** | Department | | |
| Description | | DID Number | | |
| Language | Default Language | Transmitted CID | | |
| Feature Profile | 0 - System | - E911 CID | | |
| 🗖 Enable Dial-By-Name 🛛 Enable Intercom 🗖 Agent 🌂 🗹 Release SIP Tie-Link Trunk | | | | |

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.

| | Other Call Restrictions | |
|---|---|--|
| | Allow Calls to be Transferred or Conferenced to an Outside Number | |
| 4 | Allow Extension User to Configure Forwarding, Notification and Reminder Call to an Outside Number | |
| | Allow Outside Caller to Make or Return Calls from within VM System | |
| | Allow Outside Caller to Make or Forward International Calls from within VM System | |
| | | |

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.



| General Gro | up 📔 Speed Dialing | |
|---|--------------------|--|
| Restriction | Answering | |
| Forward All Calls Finable Forward to Free Format | | |
| 915102520001, | | |
| | | |

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.

| Forward All Calls |
|-------------------------------|
| Enable Forward to Free Format |
| 9 915102520001,,228 |

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.

| Forward All Calls | |
|-------------------------------|--|
| Enable Forward to Free Format | |
| | |

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.

| Configuration Maintenance Status 8 Diagnostics Scenarios Search | Configuration File |
|---|---|
| Basic • Full (Maintenance Software Update Load Auxiliary Files Software Upgrade Key Software Upgrade Wizard Configuration File | Save the INI file to the PC. Save INI File |
| | Load the INI file to the device. Browse Load INI File The device will perform a reset after loading the INI file. |

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 *nullcon-fig.ini*.
 - 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