

MAX Communication Server Release 8.5 QuickFix

AudioCodes Mediant 1000 Configuration Guide

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About This Guide

This guide describes how to configure an AudioCodes Mediant 1000 gateway for use with MAXCS 8.5.

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

- The examples in this guide illustrate the configuration for a Mediant 1000 gateway. For other models, adjust the port settings as needed.
- AltiGen does not provide general configuration support for AudioCodes devices. Contact AudioCodes support for assistance with general setup.

Refer to your gateway's product documentation, which came with your device, for installation instructions and to retrieve the IP address of the AudioCodes web configuration tool.

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 be running MAXCS Release 8.0.
- 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 that 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 6.60A.336.04. 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.

Configuring the AudioCodes Gateway

This section shows how to customize AudioCodes' software settings for interoperability with the MAX Communications Server.

Network Topology

The following figure illustrates the gateway and server deployed in the network.





Using the AudioCodes Web Configuration Tool

During these procedures, you will use the AudioCodes Web configuration tool.

The Home page shows component status. Click any component in the graphic representation of the gateway to display operational information or additional configuration options.



Figure 1: The AudioCodes Web Configuration Tool

You must stay in *Full* menu mode; *Basic* mode does not offer all of the menus and options that you will need.

If you take a break while configuring the gateway, upon return we recommend that you confirm that the menus are still set to *Full* before you continue. In some cases, switching to *Full* menu mode while on a page will not automatically refresh the page.

Use the **Burn** button to write configuration changes to flash memory.

Reset the Gateway to Default Settings

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

Reminder: Make sure that the menus set to **Full**, as shown in the next figure.

 We recommend that you save your current configuration as a precaution; click Maintenance above the menu, and then select Software Update > Configuration File. Click Save INI File and choose a name and folder.



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

Figure 2: Click Maintenance and choose Software Update > Configuration File

- 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 (*nullconfig.ini*).
 - C. Click Load INI File and follow the instructions to reboot the gateway.

Disable the DHCP Sever

Disable the DHCP server to avoid conflict with another DHCP server during the initial configuration.

- 1. Click **Configuration** above the menu. Select **Data > Data Services > DHCP Server**.
- 2. If the **WAN Ethernet** service is shown as **Enabled**, click the pencil icon on that row to edit the settings. From the menu, select **Disabled** and click **OK**.



Name	Service	Subnet Mask	Dynamic IP Range	Action
LAN switch VLAN 1	Disabled			<u>\</u>
WAN Ethernet	Disabled			<u>\</u>
Basic 🖲 Full	a 🦊 🕥	ose Connection List		
- 69				
* System				
* WOIP				
Data				
E WAN Assess				
WAN ACCESS				
Firewall and ACL				
# QoS				
■ □ ∨PN				
B Data Services				
Data Services				
DDNS				
DNS Server				
DHCP Server				
Data Routing				
Participation of Parling				
Objects and Rules				
Data System				

Figure 3: Select Data > Data Services > DHCP Server and disable WAN Ethernet

Set VOIP Parameters

Set the gateway IP address information for your system.

- 1. Select VoIP > Network > IP Interfaces Table.
- 2. Click the **0** (zero) Index button; an *Edit* button appears. Click it and configure the network settings corresponding to your network IP address scheme:
 - Set *IP Address* to the AudioCodes gateway's IP address
 - Set *Prefix length* to the subnet mask length in CIDR notation
 - Set Gateway to the default gateway's IP address
 - (Optional) DNS: Enter the primary DNS server IP address (and the secondary if relevant)
 - Leave the remaining options set to their default settings



Figure 4: Select VoIP > Network > IP Interfaces Table

3. Click **Apply**. Click **Done**. Click **Burn**. The gateway reboots and starts using the new IP address. Maintain the cabled connection between the gateway LAN port and the provisioning computer.

Set TDM Bus Parameters

The PSTN gateway's TDM bus must be configured. The parameters shown below configure the signal encoding used between the AudioCodes gateway and MaxCS.

- 1. Select VoIP > TDM > TDM Bus Settings.
- 2. Set PCM Law Select to MuLaw.

TDM Bus Settings	Basic • Full System System Network TDM DM DTDM Bus Settings Security		
🔗 PCM La	w Select	MuLaw	-
🗲 TDM BI	us Clock Source	Network	-
🗲 TDM B	us PSTN Auto FallBack Clock	Disable	•
🗲 TDM B	us PSTN Auto Clock Reverting	Disable	•
🗲 Idle PC	CM Pattern	255	
🗲 Idle AB	3CD Pattern	0x0F	•
TDM B	us Local Reference	1	
🗲 TDM B	us Type	 Framers	-

Figure 5: Select VoIP > TDM > TDM Bus Settings and set PCM Law Select to MuLaw

3. Click Submit.



Set PSTN Trunk Parameters

Important! In this example, we use the typical PRI parameters. Before you configure this section, talk to your PRI service provider to get the correct protocol type and parameters.

- 1. Select VoIP > PSTN > Trunk Settings.
- 2. Set these parameters:
 - Set Protocol Type to T1 NI2 ISDN
 - Set Framing Method to T1 FRAMING ESF CRC6
- 3. Click Submit.

	Trunk Settings		
Basic 🖣 Full	0		
System VoIP	General Settings		
* Network	Module ID	1	
C TDM	Trunk ID	1	
* Security	Trunk Configuration State	Active	
CAS State Machines	Protocol Type	T1 NI2 ISDN	
Trunk Settings	 Trunk Configuration 		
* Signaling	Clock Master	Recovered *	
Media	Auto Clock Trunk Priority	0	
Applications Enabling	Line Code	88ZS 💌	
Control Network	Line Build Out Loss	0 dB	
* SIP Definitions	Trace Level	No Trace	
Coders And Profiles	Line Build Out Overwrite	OFF ¥	
GW and IP to IP	Framing Method	T1 FRAMING ESF CRC6	

Figure 6: Select VoIP > PSTN > Trunk Settings and adjust the settings

Set IPMedia Parameters

Configure the IPMedia parameters.

- 1. Select VoIP > Media > IPMedia Settings.
- 2. Set these (optional) parameters:
 - Set IPMedia Detectors to Enable
 - Set *Enable AGC* to **Enable**
- 3. Click Submit.



	PMedia Settings		
	IPMedia Detectors	Enable	-
C Barrie C Full	Enable Answer Detector	Disable	-
basic Full	Answer Detector Activity Delay	0	
* System	Answer Detector Silence Time	10	
VoIP	Answer Detector Redirection	0	
* Network	Answer Detector Sensitivity	0	
B Security	Answer Machine Detector Sensitivity Parameter Suit	0	•
* PSTN	Answer Machine Detector Sensitivity	3	
* Signaling	Answer Machine Detector Beep Detection Timeout	200	
B/ Media	Answer Machine Detector Beep Detection Sensitivity	0	
Voice Settings	Enable AGC	Enable	•
RTP/RTCP Settings	AGC Slope	3	
IPMedia Settings	AGC Redirection	0	•
General Media Settings	AGC Target Energy	19	
Analog Settings	Enable Energy Detector	Disable	•
Media Realm Configuration	Energy Detector Quality Factor	4	
_ imedia Security	Energy Detector Threshold	3	
	Enable Pattern Detector	Disable	*

Figure 7: Select VoIP > Media > IP Media Settings and adjust the parameters

Set SIP General Parameters

Configure the gateway SIP parameters.

- 1. Select VoIP > SIP Definitions > General Parameters.
- 2. Set the following parameters:
 - Set Enable Early Media to Enable
 - Set Session Expires Time to 300
 - Make sure that SIP Transport Type is set to UDP
 - Make sure that SIP TLS Local Port is set to 5061

3. Click Submit.



SIP General SIP General SIP General Security Secu
Advanced Parameters SIP UCP Local Port 5060

Figure 8: Select VoIP> SIP Definitions > General Parameters

Set Proxy and Registration Parameters

Configure proxy and registration parameters.

- 1. Choose VoIP > SIP Definitions > Proxy & Registration.
- 2. Set Use Default Proxy to Yes.
- 3. Click the arrow below the *Use Default Proxy* option to open the Proxy Set Table.

	Proxy & Registration		
Basic 🗘 Full	0.		
* ☐ System ■ ↓ VoIP ● @ Network	Use Default Proxy Proxy Set Table Proxy Name	Yes	¥
* Security	Redundancy Mode Proxy IP List Refresh Time	Parking 60	
* PSTN * Signaling	Enable Fallback to Routing Table Prefer Routing Table	Disable No	•
Services Applications Enabling	Use Routing Table for Host Names and Profiles Always Use Proxy	Disable Disable	•
Control Network	Redundant Routing Mode SIP ReRouting Mode	Proxy Standard Mode	•
General Parameters	Enable Registration Registration Time	Disable 180	•
Proxy & Registration	Re-registration Timing [%] Registration Retry Time	50 30	

Figure 9: Select VoIP > SIP Definitions > Proxy & Registration



4. On row 1, set *Proxy Address* to the MaxCS server's IP address and port 5060.

Default Proxy Sets Tabl	e					
	•					
	Proxy Set	ID	0		•	
			Proxy Address		Transport Type	
		1	10.30.9.203:5060		UDP V	
		2			T	
		3			•	
		4			•	
		5			•	
				· · ·		-
	-					
	Enable Pro	оху К	eep Alive Using	g Options	T	2
	Proxy Kee	ep Aliv	ve Time 60			
	Proxy Loa	d Bal	ancing Method Disat	ble	T	
	Is Proxy H	lot S	No No		T	

Figure 10: For Proxy Set ID 0, configure row 1 parameters

- 5. Set *Transport Type* to **UDP**.
- 6. Set *Enable Proxy Keep Alive* to **Using Options**.
- 7. Click Submit.

Set VoIP Coder Parameters

Configure the VoIP coder parameters.

- 1. Choose VoIP > Coders and Profiles > Coders. Set the following parameters:
 - For *Coder Name*, select the codec type to **G.711 U-la**w.
 - Set Silence Suppression to Disable.
- 2. Click Submit.



Basic ® Full	oders Table								
E System									
Rentwork	Coder Nam	ie	Packetizat	tion Time	Ra	te	Payload Type	Silence Supp	ression
* TDM	G.711U-law	•	20	•	64	•	0	Disabled	•
* Security		۲		•		•			•
* PSTN Right Andre		•		•		•			•
* Services		•		۲		•			۲
Applications Enabling		۲		•		•			•
Control Network		۲		۲		•			۲
Coders and Profiles		۲		۲		•			•
Coders		۲		۲		•			•
Coders Group Settings		۲		۲		•			•
Let Prome Settings		•		•		•			•

Figure 11: Select VoIP > Coders and Profiles > Coders

Set VoIP Trunk Group Parameters

Configure VoIP Trunk group settings.

- 1. Choose VoIP > GW and IP to IP > Trunk Group > Trunk Group.
- 2. For each trunk group, set these parameters:
 - Set *Module* to the module number and type (e.g., PRI) on which the trunks are located
 - Set From Trunk (first available) and To Trunk (last) to the physical trunk range
 - For *Channels*, specify **1-23**
 - Set *Phone Number* to the number for the first channel; the rest will be assigned sequentially
 - Set Trunk Group ID to the ID group

⊖ Basic ● Full	\odot								
⊕									
□@VoIP									
Metwork	Trunk Gro	oup Table							
€@том	-								
* Becurity	Ada	Phone Context A	e Prefix			Disable	-		
			STICILX				•		
	Tru	nk Group Index				1-10	•		
Applications Enabling	Group	Module	From	To	Channels	Phone Number	Tru	nk Group	Tel Profile ID
± Control Network	Index		Trunk	Trunk				ID	
± SIP Definitions	1	Module 1 PRI 👻	1 🔻	2 🔻	1-23	4089779200	1		0
± [™] Coders and Profiles	_								
GW and IP to IP	2	· · · · · ·							
E Trunk Group	3		-	-					
Trunk Group									
Trunk Group Settings	4	-							
	5	_		-					
			L		L	L			

Figure 12: Select VoIP > GW and IP to IP > Trunk Group > Trunk Group

3. Click Submit.



Set VOIP Trunk Group Parameters

Configure the Trunk Group setting.

- 1. Select VoIP > GW and IP to IP > Trunk Group > Trunk Group Settings.
- 2. For each trunk group, set *Channel Select Mode* to **Descending** and click **Submit**.

C Basic 🤆 Full	(\bigcirc						
* System		1						
BUDVolP								
* Network	Touch	0	oun Setting					
€@TDM			oup second	-				
* Security								
PSTN		1	•					
			Index			1-10 💌		
Services		_						
Applications Enabling			Trunk	Channel Select Mode	Registration	Serving IP	Gateway Name	Contact User
Control Network			Group G		Mode	Group 10		
SIP Definitions		1		Descending	. <u> </u>	-	I	
Coders And Profiles		2						
GW and IP to IP		-						
Garage Ga		3		·				
Garrunk Group		3						
Trunk Group		3 4						

Figure 13: Select VoIP > GW and IP to IP > Trunk Group > Trunk Group Settings

Configure IP to Trunk Group Routing

Important! Be sure to set the source IP address as the MaxCS server's address instead of a wild card; otherwise, a device from any IP address can use the gateway to make PSTN calls.

- 1. Choose VoIP > GW and IP to IP > Routing > IP to Trunk Group Routing.
- 2. Set *Source IP Address* to the IP address of the MaxCS server.
- 3. Click Submit.

	1.20					Advanced Paramet
	Routing Index	1-12	•			
	IP To Tel Routing Mode	Route cr	ils before manipulation 🔻		_	
Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Source SRD ID	->	Trunk Group ID	Source IP Group
•		10.30.9.203	-1		1	1
			-1			
1			1 4			

Figure 14: Select VoIP > GW and IP to IP > Routing > IP to Trunk Group Routing

Set Digital Gateway Parameters

Configure digital gateway options.

- 1. Choose VoIP > GW and IP to IP > Digital Gateway > Digital Gateway Parameters.
- 2. Set *B-Channel Negotiation* to **Preferred**.
- 3. Click Submit.



Digita	al Ga	teway Parameters		
C Basic @ Full	2			
System VoIP VoIP Network TDM Security	•	B-channel Negotiation Swap Redirect and Called Numbers MFC R2 Category	Preferred No	• 🖉
Becality Becality Big PSTN Big PSTN Signaling		Disconnect Call on Busy Tone Detection (CAS) Disconnect Call on Busy Tone Detection (ISDN)	Disable	•
Media Services	ľ	Enable TDM Tunneling Send Screening Indicator to IP	Not Configured	* *
Applications Enabling Control Network SIP Definitions	Г	Add IE in SETUP		
Coders And Profiles GW and IP to IP	Ŀ	Frunk Groups to Send IE Enable User-to-User IE for Tel to IP	Disable	-
Manipulations	Г	Enable User-to-user IE for IP to Tel Enable ISDN Tunneling Tel to IP	Disable Disable	•
Country Country Analog Gateway	Ŀ	Enable ISDN Tunneling IP to Tel	Disable	
Digital Gateway	L	Remove CLI when Restricted	No	-
ISDN Supp Services		Tdm Over IP Minimum Calls For Trunk Activation	0	_

Figure 15: Select VoIP > GW and IP to IP > Digital Gateway > Digital Gateway Parameters

4. Click **Burn** and reboot the gateway.

Configuring the MAX Communications Server

This section shows how to configure MaxCS to work with the AudioCodes gateway.

Configure the SIP Trunk in Enterprise Manager

Configure the SIP trunk in Enterprise Manager.

- 1. In MaxAdministrator, select **VoIP > Enterprise Network Management**.
- 2. In Enterprise Manager, click the **Codec** button on the top menu bar.

	ger (Lodec)			211
Login Logout	Password Serve	rs Codec User Departme	nt Global LCR Help At	oout
	Profile Setting			
	Name:	Mediant1000		
	Codec:	Selected Codec G.722 G.711 Mu-Law	< Add Remove> Up Down	Available Codec G.711 A-Law G.723.1 G.729 Advanced
	DTMF Delivery	RFC 2833		

Figure 16: In Enterprise Manager, click the Codec button at the top

- 3. In the lower left corner, click **Add** to add a new codec profile. Name the profile **Mediant1000** and assign the following properties:
 - For Selected Codec, select the appropriate codec and then add G.711 Mu-Law
 - Set *DTMF Delivery* to **RFC 2833**

ALTIGEN

- Set SIP Early Media to Enable
- Set *SIP Transport* to **UDP** and click **Apply**
- 4. Click the **Servers** button on the top menu bar.
- 5. Click the **IP Codec** tab, and in the IP Codec pane, click **Add** (the button below the IP Device Range).



Device Range	· ·	Ý		· ·	
From	т	To and the second se	Codec		Pipe
3 20 0 68		Newice Departs		Local	
0.20.0.108	Add IP U	Device Range		Local	
0.20.0.111	10.20.0			Local	
0.30.10.39	18.30 1 Example	10 20 6 55		Local	
0.40.0.127	From:	10.30.6.55		Local	
0.40.0.129	10.40.0	40.00.0.55		Local	
0.40.0.133	10.40.0 10.	10.30.0.55		Local	
0.40.0.134	10 40 0 Ooder	Mediant1000		Local	
0.40.1.11	Coder	. (Mediant 1000		Local	
0.140.0.20	10.140			Local	
0.140.0.43	10.140			Local	
0.104.176.40	59.104			NA	
5 254 44 194	65 254 11191			NA	
4.01.71.10	74.01.75.10	SIP.US		NA	

Figure 17: In Enterprise Manager, click Add and enter the IP device range

- 6. Enter the gateway IP address for both the *From* and *To* IP addresses.
- 7. For *Codec*, select Mediant1000.
- 8. Click **OK**.

Configure the SIP Group in MaxAdministrator

1. First, create a SIP Group. In MAXCS Administrator, double-click **SIPSP** in *Boards* view. Click **Board Configuration** and then click **SIP Group Configuration**.

	SIP Signaling Channel Configuration
Boards Image: Constraint of the second sec	SIP Extension Channels Current Configured Channels Change Number of SIP Extension Channels to
Channel Mapping List Ugical Type Physical Channel Mapping List Ugical SIP Extension Reset Channel Board Configuration Reset Board	SIP Tie-Trunk Channels (Connecting AltiServ-to-AltiServ VoIP calls) Current Configured Channels T2 Change Number of SIP Tie-Trunk Channels to SIP Trunking Channels (Connecting 3rd party SIP Dial Tone to AltiServ) Current Configured Channels Current Configured Channels Current Configured Channels Change Number of SIP Trunk Channels to SIP Group Configuration Channel Assignment Advanced Configuration "Note: Changing number of SIP extension or tie trunk channels requires stop and re-start switching and gateway services.
	OK Cancel

Figure 18: Open the SIP Group Configuration panel

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2. Click **Add**. Enter a name for the group (for example, *M1000*). Do not check the *AltiGen Trunk* option. Click **OK**.

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

Figure 19: Enter a name for this new SIP Group

3. 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 *M1000*) and click the lower **Add** button (the one that is below the *SIP Servers* list). Enter the URL for the domain and click **OK**.

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

Figure 20: Add a SIP server to the new SIP Group

- 4. Highlight the new SIP Group. Select this new SIP Server and configure the settings on the *Register* tab.
 - Enter the SIP Server IP Address in the second field.
 - For User Name, enter audiocodes
 - Leave the *Password* field empty; AudioCodes does not accept registration, so no password is needed.



- For the Register Period, enter **0**
- For the SIP Source Port (Non-TLS) and the SIP Destination Port options, enter 5060

Register Settings SIP OPT	IONS
Domain:	10.30.6.55
SIP Server IP Address:	10.30.6.55
User Name:	audiocodes
Password:	
Register Period:	0
SIP Source Port (Non-TLS):	5060
SIP Destination Port:	5060

Figure 21: Configure SIP Trunk parameters

- 5. Switch to the *Settings* tab and set the following parameters:
 - Set the SIP Protocol Field to From Header
 - Set the options for the carrier's ability to accept transmitted caller ID based on the restrictions provided by the PRI carrier. In this example, *Carrier can only accept Calling Number with minimum xx digits* is set to **10** digits.
 - For Use this as Calling Number if the Carrier cannot accept configured numbers, enter your main AudioCodes number.
 - Select **Send Caller Name** to send the extension's name in the outbound call. Note that many remote systems may not receive or display this information.
 - Make sure that *Enable Standard Record-Route Header* is NOT checked.
 - Select **To Header** for the *Incoming DID Number* field.
- 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.



Ch	annel Assign	ment			×
[Enabled	ID	Channel No	Group	Max SIP Trunk License
		0	72	M1000TLS	32
		1	73	M1000TLS	
		2	74	M1000TLS	Assigned SIP Trunk License
		3	75	M1000TLS	15
		4	76	M1000TLS	J
		5	77	M1000TLS	Assign Group
		6	78	M1000TLS	
		7	79	M1000TLS	If you need more channels, close this
		8	80	M1000TLS	panel, click the 'Board Configuration'
		9	81	M1000TLS	for "Change the Number of SIP Trunk
		10	82	M1000TLS	Channels to". Then restart the
		11	83	M1000TLS	system.
		12	84	M1000TLS	
		13	85	M1000TLS	
		14	86	M1000TLS	
		15	87		
		16	88		
		17	89		
		18	90		
		19	91		

Figure 22: Assign channels to the new group and enable those channels

- 7. Add a trunk access code to the SIP Trunks that you just configured.
- 8. Close all of the SIP Group/Board Configuration windows.
- 9. Re-open *Boards* view. Double click **SIPSP**, select **Board Configuration**, and click **Advanced Configuration**. Add the gateway's IP address, 10.30.6.55, to the *Trusted SIP Device* List.

SIP Device IP Address	
10.20.0.125	
10.20.0.136	
10.20.0.141	
10.30.5.115	10
10.30.6.55	
10.30.6.201	
10 20 9 129	
10 30 8 199	
10.30.9.33	
10.30.9.99	
10.30.9.170	
10.30.9.253	
10.30.10.11	
10.30.10.13	
10.30.10.16	
10.30.10.17	
10.30.10.18	

Figure 23: Add the gateway's 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.



10. Verify the configuration by making inbound and outbound calls. Do not proceed any further with this configuration until you can successfully make and receive calls.

Configuring PRI TBCT - Release Link Tie for PRI (Optional)

Before you begin these procedures, be aware that you will be stopping the trunk, which will cause any calls over PRI to drop. Consider performing this step during a period of low call volume.

- 1. Call your PRI provider to check if they support PRI TBCT (Two B Channel Transfer). If they do, ask them to enable it. Also, please be aware that additional fees may be charged for every TBCT that is made from your PRI provider. Check with your PRI provider to determine if any additional fees will be incurred.
- 2. Make sure you can make inbound and outbound regular PRI calls without issues. If you cannot make regular inbound or outbound calls, repeat the previous sections until you resolve the issue.
- 3. In the AudioCodes configuration tool, make sure the menu option **Full** is selected.



Figure 24: Make sure the Menu is set to Full

 Select Configuration > VoIP > PSTN > Trunk Settings. Click Stop Trunk. (Warning: Any active calls on the PRI lines will be dropped!)



5. Change *Transfer Mode* to **TBCT**.

•		
PSTN Alert Timeout	-1	
Transfer Mode	ТВСТ	~
Local ISDN Ringback Tone Source	PBX	~
Set PI in Rx Disconnect Message	Not Configured	\checkmark

Figure 25: Change Transfer Mode and click Apply Trunk Settings

- 6. Click **Apply Trunk Settings** near the bottom-left corner to re-enable the trunk.
- 7. In MaxCS Administrator, open *Boards* view. Double-click **SIPSP**, click **Board Configuration**, and then click **SIP Group Configuration**.



8. Select the SIP group in the list, and then select the SIP server below. Switch to the *Settings* tab and check **Enable SIP REFER**.



Figure 26: Check the Enable SIP REFER option

9. In MaxCS Administrator, create a virtual extension. The next figure shows a virtual extension 400. Check the **Release SIP Tie-Link Trunk** option.

Num Type 240 Virtual 250 IP(Agent) 251 IP(Agent) 252 IP(Agent) 253 IP(Agent) 254 Virtual 255 Virtual(A, 256 IP 260 Virtual 270 Virtual 287 Virtual 400 Virtual	Name ip talkzero gold zero gold 2 gold 2 gold 3 gold 4 gold 5 remote a audio1 test TBCT D	Hestriction General General Personal Info First Name Password Description Language Feature Prol Enable I Account Cod	n Answering Answering Group Speed Di Imation TBCT DEMO	Une Numbe ialing Ma Last Name Department DID Number Transmitted CID E911 CID Segnt Recording Options	r Access iil Management III Release SIP	Monitor List Notification Tie-Link Trunk
General	Group	Speed Dialin	a Î Mail Manageme	ent Í N	otification	
Restriction		Answering 1	One Number Access	AN I N Mor	nitor List	
C Allow Inter	nal/Local/Unres	stricted Area Codes estricted, and defined p	orefixes			
C Allow Internal, LC C Allow Intern C All calls allo Prefixes Allowed	nal/Local/Unres	stricted Area Codes stricted, and defined p e defined prefixes	Prefixes			
C Allow Inter C All calls alk Prefixes Allowed Enter prefix less 1900 or 976)	ical, and Unres hal/Local/Unre wwed except th	stricted Area Codes estricted, and defined p e defined prefixes	Prefixes Prefixes Disallowed Enter prefix less or equal or 976)	to 10 digits. (e.ç		
C Allow Inter C All calls alk Prefixes Allowed Enter prefix less 1900 or 976) Other Call Restr	ical, and Unres hal/Local/Unre bwed except th sor equal to 10 ictions	e defined prefixes	Prefixes Prefixes Disallowed Prefixes Disallowed Enter prefix less or equal or 976) an Outside Number	to 10 digits. (e.ç	J. 1900	

Figure 27: Create a new virtual extension



- 10. For the same virtual extension, open the extension's *Restriction* tab and check the option Allow Extension User to Configure Forwarding, Notification and Reminder Call to an Outside number. In this example, the option No Restriction on Outcalls is also selected.
- 11. On the extension's Answering tab, select *Enable Forward to* and choose **Outside Number**. Enter a target number. In the next example, 14085979000 is the target number.

1	General	Group	1	Speed Dialing	
	Restriction		Answeri	ng	
	- Forward All Call	ls			
Z	🔽 Enable Fo	rward to Ou	itside Num	iber 💌	
	9 14085979000				

Figure 28: Select the Enable Forward option and choose Outside Number

To verify the configuration,

- 1. Make a call from your cell phone to the MaxCS system, the number is 14085979200 in this example. The call will enter auto attendant.
- 2. Dial 400, the call will be transferred to 14085979000. Check the MaxCS Administrator Trunk View; initially, it will use two SIP trunks, but once the call is connected, two of the trunks will be released shortly. The voice between your cell phone and 4085979000 will remain.
- 3. Remove the PRI cable from the gateway. The voice path between your cell phone and the call (408597900) will remain. (Be aware that the other active calls on the PRI lines will drop during this testing.)

Configuring TLS Support (Optional)

This section shows how to configure the gateway for TLS support.

Set Web Security Settings

AudioCodes' SIP TLS cannot work properly if Web Management's HTTPS is not turned on properly.



THESE STEPS ARE IMPORTANT – If you set the Web Security Settings incorrectly, then you may need to reset your device to its initial defaults and reconfigure it again from scratch.

- 1. To configure TLS, select **System > Management > WEB Security settings.**
- 2. Set *Secured Web Connection (HTTPS)* to **HTTP and HTTPS**. (If you set this to only HTTPS and the device stops working, you may need to reset the device and reconfigure it again.)
- 3. Set *HTTPS Cipher String* to **DEFAULT**. This field is case-sensitive; **your entry must be in all uppercase letters** as shown in the following figure.



Configuration Maintenance Status 8 Diagnostics Search	Web Security Settings		
▼ General			
Basic 🖲 Full	Voice Menu Password	12345	
□@_System	Secured Web Connection (HTTPS)	HTTP and HTTPS 🗸	
Application Settings	Requires Client Certificates for HTTPS connection	Disable 🗸 🗸	
Syslog Settings	🔗 HTTPS Cipher String	DEFAULT	
Regional Settings	🔗 WAN OAMP Interface	Not Configured 🗸	
Certificates	Allow WAN access to HTTP	Disable 🗸	
Management	Allow WAN access to HTTPS	Disable 🗸	
WEB Security Settings	✓ Session		
Telnet/SSH Settings	Session Timeout (minutes)	15	

4. Use the **Burn** button to write configuration changes to memory, and then reboot the gateway.

Set Proxy & Registration Settings

Set the proxy and registration parameters.

- 1. In the AudioCodes web utility, choose **VoIP** > **SIP Definitions** > **Proxy & Registration**.
- 2. Set Use Default Proxy to Yes.
- 3. Click the arrow below the *Use Default Proxy* parameter to open the Proxy Set Table.

Basic (* Full	() ·		
	Use Default Proxy	Yes	
VoIP	Proxy Set Table		
* Network	Proxy Name		
*@TDM	Redundancy Mode	Parking	*
Becurity	Proxy IP List Refresh Time	60	
PSTN Signaling Applications Enabling Control Network SIP Definitions General Parameters Advanced Parameters	Enable Fallback to Routing Table	Disable	
	Prefer Routing Table	No	-
	Use Routing Table for Host Names and Profiles	Disable	
	Always Use Proxy	Disable	
	Redundant Routing Mode	Proxy	•
	SIP ReRouting Mode	Standard Mode	
	Enable Registration	Disable	
	Registration Time	180	
Provy & Registration	Re-registration Timing [%]	50	
E Floxy & Registration	Registration Retry Time	30	

Figure 29: Select VoIP > SIP Definitions > Proxy & Registration

- 4. In row 1, set *Proxy Address* to the MaxCS server's IP address and port 5061.
 - Set *Transport Type* to **TLS**
 - Set Enable Proxy Keep Alive to Using Options



5. Click Submit. Click Burn.

Default Proxy Sets Tabl	e				
	•				
	Proxy Set ID		0	•	
		Proxy Addres	5	Transport Type	1
	1	10.30.9.203:5061		TLS 🔻	1
	2			•	1
	3			•	1
	4			•	1
	5			•	1
					-
	-				
	Enable Proxy K	(eep Alive	Using Options	•	0
	Proxy Keep Ali	ve Time	60		
	Proxy Load Ba	lancing Method	Disable	•	
	Is Proxy Hot S	wap	No	•	

Figure 30: Configure Proxy Set ID 0 parameters

6. Select **VoIP** > **Media** > **Media Security**. Change *Media Security* to **Enable**.

Mediant Mediant	1000 - MSBG Submit O Burn Device Ad	tions 🔻 💼 Home 🙆 Hel
Configuration Maintenance Status & Diagnostics	Media Security	
Search	General Media Security Settings	
Basic O Full	Media Security	Enable
€ 🗇 System	Media Security Behavior	Preferable •
DIP	SRTP Tunneling Authentication for RTP	Disable •
	SRTP Tunneling Authentication for RTCP	Disable •
€@TDM		
• Security	✓ SRTP Setting	
	Master Key Identifier (MKI) Size	0
Planedia	Symmetric MKI Negotiation	Disable •
Voice Settings Fax/Modem/CID Settings RTP/RTCP Settings General Media Settings Media Realm Table Media Security		

Figure 31: Select VoIP > Media > and set Media Security to enable

7. Burn, and then reboot the Gateway.

Configure TLS in the Codec Profile in Enterprise Manager

Create a new codec profile in Enterprise Manager.

- 1. In MaxCS Administrator, select VoIP > Enterprise Network Management.
- 2. In Enterprise Manager, click the **Codec** button on the top menu bar.



- 3. In the lower corner, click **Add** to add a new codec profile. Configure these properties:
 - Assign the name Mediant1000TLS
 - Select the appropriate codec and then add G.711 Mu-Law
 - Set *DTMF Delivery* to **RFC 2833**
 - Set SIP Early Media to Enable
 - Set SIP Transport to TLS

and the second						
Login Logout Password	Servers Codec User Department C	Biobal LCR Help About				
Profile Setting						
Name:	Name: Mediant1000TLS					
Codec:	Selected Codec G.722 G.711 Mu-Law	Available Codec < Add G.711 A-Law G.723.1 G.729 Up Down				
		Advanced				
DTMF Deliv	ery (RFC 2833	•				
SIP Early M	edia Enable	•				
SIP Transp	ort (TLS	•				

Figure 32: Open Enterprise Manager and add a new codec

- 4. Click the **Servers** button. Click the **IP Codec** tab.
- 5. In the IP Codec pane, select the entry points to Mediant 1000 and click Edit.
 - Enter the gateway IP address for both the *From* and *To* IP addresses
 - For Codec, select Mediant1000TLS
- 6. Click **OK**.



P Device Range					
From	То	ĺ.	Codec	1	Pipe
0.20.0.68	10 20 0 CT 1 1 10 0			Local	
0.20.0.108	Add IP Dev	vice Range	×	Local	
0.20.0.111	10.20.0			Local	
10.30.10.39	10.30 1 Erom:	10 20 6 55		Local	
0.40.0.127	10.40.0 FIGHL	10.00.0.00		Local	
0.40.0.129	10.40.0 To: C	10 20 6 55		Local	
10.40.0.133	10.40.0	10.30.0.35		Local	
0.40.0.134	Codec	Codec: Mediant1000TLS		Local	
0.401.11	Codec.	Mediantrooores		Local	
10.140.0.20	10.140	Cancel		Local	
0.140.0.43	10.140/			Local	
59 104 176 40	58.104			NA	
15.254.44.194	85.254 <u>- 1.1 9 1</u>	011.00		NA	
14.01.71.10	74.01.75.10	SIP.US		NA	

Figure 33: Define the range for the new IP Device

Configure the SIP Trunk in MaxAdministrator

Set up the SIP Trunk in MaxCS Administrator.

1. In MAXCS Administrator, double-click **SIPSP** in *Boards* view. Click **Board Configuration** and then click **SIP Group Configuration**.

	SIP Signaling Channel Configuration
Boards Logic Board Type Board Configuration Board Info 1 MobileExtSP 2 SIPSP Board Logical ID Board Name SIPSP-0@GW	SIP Extension Channels Current Configured Channels Change Number of SIP Extension Channels to
Channel Mapping List Channel Mapping List O SIP Extension	SIP Tie-Trunk Channels (Connecting AltiServ-to-AltiServ VoIP calls) Current Configured Channels 12 Change Number of SIP Tie-Trunk Channels to 12 SIP Trunking Channels (Connecting 3rd party SIP Dial Tone to AltiServ) Current Configured Channels 16 Change Number of SIP Trunk Channels to 16 SIP Group Configuration Channel Assignment Advanced Configuration Advanced configuration *Note: Changing number of SIP extension or tie trunk channels requires stop and re-start switching and gateway services.
	OK

Figure 34: In MaxCS Administrator, open the SIP Group Configuration panel



- 2. In the SIP Trunk Configuration window, select the SIP Group that you created earlier (*M1000*) and the SIP Server. Check the following parameters:
 - The SIP Server IP Address should be the gateway IP address
 - The User Name should still be AudioCodes
 - Leave the password empty
 - The SIP Register Period should be 0
 - The *SIP Source Port* should be 5060
 - Set the SIP Destination Port to 5061
- 3. Next, check that the channels are enabled (you did this step earlier). In the Board Configuration panel. Click **Channel Assignment**. Confirm that the channels are assigned and are enabled.
- 4. Add a trunk access code to the SIP Trunk that you just configured.
- 5. Click **OK** to save the changes.
- 6. Click **OK** again to close the *SIP Trunk Configuration* window.
- 7. Verify the configuration by making inbound and outbound calls.

Troubleshooting

If making outbound PSTN calls sometimes fails, open the AudioCodes configuration tool, select VoIP > GW and IP to IP > Digital Gateway and set *Remove Calling Name* to Enable.