

MAX Communication Server Release 8.0

AudioCodes MP-11x Configuration Guide

Intended audience: AltiGen Authorized Partners

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About This Guide

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

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

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

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

Requirements

Your system and environment must meet the following requirements:

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

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



Overview

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

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



Figure 1: Example configurations for Internet and LAN deployments

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

General Configuration

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

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

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



3. Add a new profile named *audiocodes*.

Name:	audiocodes	
	Selected Codec	
	🗲 G.711 Mu-Law	< Add
		Remove
Codec:		Up
		Down
DTMF Delivery	RFC 2833	
CID Fork Modi	Enable	

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

Figure 2: Add a new Codec Profile

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

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

🚔 Add IP Device Range	×
From: 10.40.0.95	
To: 10.40.0.95	
Codec: audiocodes	-
OK Cancel	

Figure 3: Add the Firewall IP address range

- Set both the *From* and the *To* fields to 10.40.0.95 (the firewall IP address shown in the example on page 4) or the gateway's IP address
- For the Codec, select audiocodes
- 5. Open the MaxCS Administrator *Boards* view. Double click **SIPSP**, select **Board Configuration**, and click **Ad**-**vanced Configuration**. Add **10.40.0.95** to the *Trusted SIP Device List*.



٩dva	anced Configuration
⊢ ⊺	rusted SIP Device List
	SIP Device IP Address
	10.40.0.57
	10.40.0.67
	10.40.0.70
	10.40.0.95
	177.28.234.34

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

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

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

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

Important!While working in the AudioCodes configuration application, make sure that Full is always se-
lected for the *Configuration* menu, so that you can see all of the menu options. Check this set-
ting if you are idle for a few minutes; sometimes the application resets the menus to Basic
mode.



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

Figure 5: Set the AudioCodes tool menu to Full

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



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

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

C Basic 📀 Full		\bigcirc			
€ System ■ System ■ System ■ Systematic					
IP Interfaces Table		IP Ro	uting Table		
IP Routing Table	#	Delete Row	Destination IP Address	Prefix Length	Gateway IP Address
QoS Settings	1	Γ	192.168.1.20	24	192.168.1.1
	2				
	3	Г			

Figure 6: Confirm the network settings

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



Figure 7: Configure the TX and RX Payload Type settings



10. Select VoIP > Media > General Media Settings.



Figure 8: Disable NAT Traversal

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

Basic • Full System VoIP ColP Co	(3	 Set Proxy Address MAXCS public IP a Transport Type to Set Enable Proxy a you configure sur will change this se Set Proxy Keep Al 	5 1 to 10 address o UDP . <i>Keep Al</i> vival se etting.) <i>ive Tim</i>	0.40.1.43:10060 (The and port). Set its <i>live</i> to Disable . (If later ettings on FXS ports, you <i>e</i> to 15 (seconds).	
Control Network	Proxy Set	t ID	→	0	v	
E PTOXY Sets Table			Proxy Address		Transport Type	
	\rightarrow	1	10.40.1.43:10060		UDP 🗸	
		2				
		3				
		4				
		5			v	
	•					
	Enable	e Pro	oxy Keep Alive 🛛 🔶	Disable	• ~	2
	Proxy	Kee	ep Alive Time 🔶	15		

Figure 9: Add an entry to the Proxy Sets table



12. Select VoIP > SIP Definitions > General Parameters.

 System 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. Set SIP Destination Port to 10060. SIP Definitions General Parameters Advanced Parameters Account Table Proxy & Registration RADIUS Accounting S Session Expires Method Fax Signaling Method Tas Relay Detect Fax on Answer Tone SIP Transport Type UDP Sip Transport Type UDP Session Expires Session Expires	🔾 Basic 🖲 Full	Make s	ure that NAT IP Address is 0.0.0.0 .
 Set SIP Transport Type to UDP. Set SIP UDP Local Port to 10060. Set SIP Destination Port to 10060. SIP General NAT IP Address NAT IP Address NAT IP Address PRACK Mode Supported Channel Select Mode By Dest Phone Number Ital Parameters Advanced Parameters Account Table Proxy & Registration RADIUS Accounting S Session Expires Method Fax Signaling Method Tas Relay Detect Fax on Answer Tone Sip Transport Type UDP SIP UDP Local Port Sofo1 Sip TCP Local Port Sofo1 Enable SIPS Disable Disable Sip TCP Local Port Sofo1 Control Sip Poetination Por	± 🗐 System	Set Ena	<i>ble Early Media</i> to Disable .
 Network Security Media Services Applications Enabling Control Network SIP Definitions General Parameters Advanced Parameters Account Table Proxy & Registration RADIUS Accounting S Sester I dentity Mode Session-Expires Solo Session-Expires Solo Session-Expires Solo Session-Expires Session-Expir	□ 🖾 VoIP	Set SIP	Transport Type to UDP .
 Set SIP Destination Port to 10060. Services Applications Enabling NAT IP Address National Select Mode September Advanced Session-Expires Session-Expires<th></th><th>• Set SIP</th><th>UDP Local Port to 10060.</th>		• Set SIP	UDP Local Port to 10060 .
 Services SIP General NAT IP Address O.0.0 PRACK Mode Supported PRACK Mode By Dest Phone Number Channel Select Mode By Dest Phone Number Session-Expires Session-Expires Session-Expires Session Expires Method Fax Signaling Method T.38 Relay Detect Fax on Answer Tone SIP Transport Type UDP SIP UDP Local Port SiP UDP Local Port SiP TLS Local Port SiP TLS Local Port SiP TLS Local Port Sipable TCP Timeout SIP Destination Port 	⊕@_Media	• Set SIP	Destination Port to 10060 .
 Applications Enabling Applications Enabling Control Network SIP Definitions General Parameters Advanced Parameters Advanced Parameters Account Table Proxy & Registration RADIUS Accounting S Sip Transport Type Disable SIP Transport Type UDP SIP Transport Type UDP SIP Transport Type SIP Transport Type Sip Transport Type Sip Transport Type Sip TCP Local Port Sip TCP Local Port	± 💷 Services		
Control Network PRACK Mode Supported ✓ SIP Definitions Channel Select Mode By Dest Phone Number ✓ General Parameters Enable Early Media Disable ✓ Advanced Parameters 183 Message Behavior Progress ✓ Account Table Minimum Session-Expires 90 ✓ Proxy & Registration Session Expires Method re-INVITE ✓ RADIUS Accounting S Asserted Identity Mode Disabled ✓ Fax Signaling Method T.38 Relay ✓ Detect Fax on Answer Tone Initiate T.38 on Preamble ✓ SIP UDP Local Port 10060 SIP TCP Local Port 5060 SIP TLS Local Port 5061 Enable SIPS Enable ✓ Enable SIPS Disable ✓ SIP TCP Local Port 5061 SIP Destination Port 0 SIP Destination Port 0	Emplications Enabling	🗲 NAT IP Address	
SIP Definitions Channel Select Mode By Dest Phone Number General Parameters Enable Early Media Disable Advanced Parameters 183 Message Behavior Progress Account Table Minimum Session-Expires 90 Proxy & Registration Session Expires Method re-INVITE RADIUS Accounting S Asserted Identity Mode Disabled Fax Signaling Method T.38 Relay V Detect Fax on Answer Tone Initiate T.38 on Preamble V SIP UDP Local Port 10060 SIP Transport Type UDP SIP TLS Local Port 5061 Enable SIPS Disable Enable SIPS Disable V Enable SIPS Disable CP Timeout 0 SIP Destination Port 10060 SIP Destination Port	🗉 🦾 Control Network	PRACK Mode	Supported 🗸
General Parameters Enable Early Media Disable ✓ Advanced Parameters 183 Message Behavior Progress ✓ Account Table Minimum Session-Expires 90 Proxy & Registration Session Expires Method re-INVITE ✓ RADIUS Accounting S Asserted Identity Mode Disabled ✓ Fax Signaling Method T.38 Relay ✓ Detect Fax on Answer Tone Initiate T.38 on Preamble ✓ SIP Transport Type UDP ✓ SIP TCP Local Port 5060 SIP TLS Local Port 5061 Enable SIPS Disable ✓ Enable TCP Connection Reuse Enable ✓ TCP Timeout 0	□ 🗐 SIP Definitions	Channel Select Mode	By Dest Phone Number
Advanced Parameters 183 Message Behavior Progress ✓ Account Table 0 Proxy & Registration RADIUS Accounting S Session-Expires 90 RADIUS Accounting S Session Expires Method re-INVITE ✓ Asserted Identity Mode Disabled ✓ Fax Signaling Method T.38 Relay ✓ Detect Fax on Answer Tone Initiate T.38 on Preamble ✓ SIP Transport Type UDP ✓ SIP UDP Local Port 5060 SIP TLS Local Port SIP TLS Local Port 5061 Enable ✓ Enable SIPS Disable ✓ Enable TCP Connection Reuse Enable ✓ TCP Timeout 0	General Parameters	Enable Early Media	Disable 🗸
Account Table O Proxy & Registration Session-Expires 90 RADIUS Accounting S Session Expires Method re-INVITE ✓ Asserted Identity Mode Disabled ✓ Fax Signaling Method T.38 Relay ✓ Detect Fax on Answer Tone Initiate T.38 on Preamble ✓ SIP Transport Type UDP ✓ SIP TCP Local Port 5060 SIP TLS Local Port SIP TLS Local Port 5061 Enable ✓ Enable SIPS Disable ✓ Enable TCP Connection Reuse Enable ✓ TCP Timeout O		183 Message Behavior	Progress 🗸
Account rable Minimum Session-Expires 90 Proxy & Registration Session Expires Method re-INVITE ✓ RADIUS Accounting S Asserted Identity Mode Disabled ✓ Fax Signaling Method T.38 Relay ✓ Detect Fax on Answer Tone Initiate T.38 on Preamble ✓ SIP Transport Type UDP ✓ SIP UDP Local Port 10060 5 SIP TLS Local Port 5061 Enable Enable SIPS Disable ✓ Enable TCP Connection Reuse Enable ✓ TCP Timeout 0 5 5 SIP Destination Port 10060 5		Session-Expires Time	0
Proxy & Registration Session Expires Method re-INVITE RADIUS Accounting S Asserted Identity Mode Disabled Image: Constraint of the second seco		Minimum Session-Expires	90
RADIUS Accounting S Asserted Identity Mode Disabled ✓ Fax Signaling Method T.38 Relay ✓ Detect Fax on Answer Tone Initiate T.38 on Preamble ✓ SIP Transport Type UDP ✓ SIP UDP Local Port 10060 5061 SIP TLS Local Port 5061 Enable Enable SIPS Disable ✓ TCP Timeout 0 51100000000000000000000000000000000000	Proxy & Registration	Session Expires Method	re-INVITE 🗸
Fax Signaling Method T.38 Relay ✓ Detect Fax on Answer Tone Initiate T.38 on Preamble ✓ SIP Transport Type UDP ✓ SIP UDP Local Port 10060 ✓ SIP TCP Local Port 5060 ✓ SIP TLS Local Port 5061 ✓ Enable SIPS Disable ✓ TCP Timeout 0 ✓	RADIUS Accounting S	Asserted Identity Mode	Disabled 🗸
Detect Fax on Answer Tone Initiate T.38 on Preamble SIP Transport Type UDP SIP UDP Local Port 10060 SIP TCP Local Port 5060 SIP TLS Local Port 5061 Enable SIPS Disable TCP Timeout 0 SIP Destination Port 10060		Fax Signaling Method	T.38 Relay
SIP Transport Type UDP SIP UDP Local Port 10060 SIP TCP Local Port 5060 SIP TLS Local Port 5061 Enable SIPS Disable Enable TCP Connection Reuse Enable TCP Timeout 0 SIP Destination Port 10060		Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP UDP Local Port 10060 SIP TCP Local Port 5060 SIP TLS Local Port 5061 Enable SIPS Disable Enable TCP Connection Reuse Enable TCP Timeout 0 SIP Destination Port 10060		SIP Transport Type	
SIP TCP Local Port 5060 SIP TLS Local Port 5061 Enable SIPS Disable Enable TCP Connection Reuse Enable TCP Timeout 0 SIP Destination Port 10060		SIP UDP Local Port	10060
SIP TLS Local Port 5061 Enable SIPS Disable Enable TCP Connection Reuse Enable TCP Timeout 0 SIP Destination Port 10060		SIP TCP Local Port	5060
Enable SIPS Disable Enable TCP Connection Reuse Enable TCP Timeout 0 SIP Destination Port 10060		SIP TLS Local Port	5061
Enable TCP Connection Reuse Enable TCP Timeout 0 SIP Destination Port 10060		Enable SIPS	Disable 🗸
TCP Timeout Image: SIP Destination Port		Enable TCP Connection Reuse	Enable
SIP Destination Port		TCP Timeout	0
		SIP Destination Port	10060

Figure 10: Configure general SIP parameters



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



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

Jse Default Proxy	Yes
Proxy Set Table	
Proxy Name	
Redundancy Mode	
Proxy IP List Refresh Time	
Enable Fallback to Routing Table	Enable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	
Always Use Proxy	Disable
Redundant Routing Mode	•
SIP ReRouting Mode	Use Routing Table
Enable Registration	Enable
Registrar Name	
Registrar IP Address	
Registrar Transport Type	UDP
Registration Time	100
Re-registration Timing [%]	50

Figure 11: Configure Proxy and Registration parameters



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

Basic Full System VoIP CoIP Security Media				
± Services	Coder Name	Packetization Time	Rate	Payload Type
Applications Enablin	G.711U-law	20 💌	64 💌	0
⊕	-			10
	G.729	20	8	18
Coders and Profiles				
Coders				
Coders Group Setti	ings			
Tel Profile Settings				
IP Profile Settings				

- Figure 12: Confirm that G.711U-law and G.729 appear in the Coders list
- 15. Select VoIP > GW and IP to IP > DTMF and Supplementary > DTMF & Dialing. You can omit the *Digit Mapping Rules* field if you are configuring only FXO.

Basic Full System VoIP ColP	 Set Max Digits In Phone Num to 20 Set Declare RFC 2833 in SDP to Yes Set 1st Tx DTMF Option to RFC 2833 Set RFC 2833 Payload Type to 101 Set Hook-Flash Option to INFO (NetCentrex) 	_
E Control Network		_
	Max Digits In Phone Num — 20	
E Coders and Profiles	Inter Digit Timeout [sec] 3	
GW and IP to IP	Declare RFC 2833 in SDP Yes	
	1st Tx DTMF Option RFC 2833	
	2nd Tx DTMF Option	
Harring	RFC 2833 Payload Type	
DIMF and Supple	Hook-Flash Option	
Supplementary	Digit Mapping Rules	

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



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

	Gateway	Caller
	Port	ID
Port 1	FXS	Enable 🗸
Port 2	FXS	Enable V
FUILZ	172	
Port 3	FXS	Enable 🗸
	EV.C	Eastle 14
Port 4	FXS	Enable V
Port 5	EXO	Enable V
FOILS	170	Enable +
Port 6	FXO	Enable 🗸
Det 7	EVO	Enable M
Port /	FXU	
Port 8	FXO	Enable 🗸

Figure 14: Enable Caller ID for FXS ports

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

System			
VoIP V	oice Settings		-
Network			
Security			
⊖@ Media	 Voice Settings 		_
Voice Settings	Voice Volume (-32 to 31 dB)	0	
Fax/Modem/CID Settin	Input Gain (-32 to 31 dB)	0	
RTP/RTCP Settings	Silence Suppression	Disable	~
IPMedia Settings	DTMF Transport Type	RFC 2833 Relay DTMF	~
	DTMF Volume (-31 to 0 dB)	-6	
	NTE Max Duration	-1	
	DTMF Generation Twist	0	
	Echo Canceller	Enable	~

Figure 15: Adjust Voice Settings

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

AudioCodes - Microsoft Internet Explorer provided by AltiGen Communications, Inc.							
G → Mathematical Mathematical Activity (10.30.15.1)	24/		<u>-</u> م				
	MP-118 FXS_FXO	Submit 🧿 Burn					

Figure 16: Click Burn to save the configuration changes

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



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

- usa_tones_13_NoHold.ini
- usa_tones_13_NoHold.dat
- **Note:** If you do not load these files, callers may hear an on-hold beep tone while calls are being connected or disconnected.



Figure 17: Upload the two data files

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

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

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

Analog Extension (FXS) Configuration

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

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



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

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



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

Figure 18: Configure Extension parameters

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



Figure 19: Enable SIP Telephony service

 In the AudioCodes configuration tool, select VoIP > Media > Analog Settings. Configure the settings as follows.

▼ FXS Settings		
🔗 Analog Metering Type	12 kHz sinusoidal bursts	~
🔗 Min. Hook-Flash Detection Period [msec]		
Max. Hook-Flash Detection Period [msec]	→ 800	
🗲 FXS Coefficient Type	USA	~

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

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

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



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



Figure 21: Configure Hunt Group Endpoint number parameters

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

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



Figure 22: Add Hunt Group ID 1



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

🔾 Basic 🖲 Full	IP To Hunt Group Routing) Table			_	
	Routing Index IP To Tel Routin	g Mode –	1-12 ∨ Route calls before ma	nipulation V		Basic Pa
Security Media	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID
Services Enabli		[167]	I <u>*</u> →	10.40.1.43		1
Control Network	ļ	ļ				
SIP Definitions	<u> </u>	<u> </u>				
• Coders and Profile	<u> </u> *			10.40.1.43		2
GW and IP to IP	<u></u>					
⊕ ☐ Hunt Group ⊕ ☐ Manipulations ⊖ ☐ Routing	•	In the upper pan manipulation.	el, set IP to Tel Rou	ting Mode to Rout	te ca	lls befo
General Parameter	rs	Prefix to an aster	isk (*).	iusi Prejix, and So	urce	FIIONE
IP to Hunt Group F	Routing g Reasons	Set <i>Destination F</i> In our example, 1 ample, [167-170]	Phone Prefix to your his is 167 . You can would include exte	FXS ports' extens enter a range if ne ensions 167 throug	ion r ede gh 1]	numbe d; for e 70.

- Set Source IP Address to the MAXCS IP address, 10.40.1.43.
- Set *Hunt Group ID* to **1**. (This assumes that the FXS port's hunt group ID is **1**.)

Figure 23: Configure IP to Hunt Group routing rule

- Note: When a SIP call is sent to this AudioCodes device, it will match rows from top to bottom. Once it detects is a match, the call will be routed the hunt group of the *first match row* (it will ignore the rest of rows). In the example in Figure 23, if you swap these two rows, then when a call is placed to extension 167, it will be sent to the FXO port instead of the FXS port.
- 8. Configure another hunt group routing entry; we recommend that you leave a few empty rows after row 1, for future expansion. In our example, we have added the new rule in row 4; refer to the preceding figure.
 - Set Destination Host Prefix, Source Host Prefix, Destination Phone Prefix, and Source Phone Prefix to an asterisk (*).
 - Set Source IP Address to the MAXCS IP address, 10.40.1.43.
 - Set Hunt Group ID to 2 (Hunt Group 2 is used for FXO in this example).

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



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

System System VoIP Security Media	 Set Enable H Set Enable T Set Enable C 	old to Disable . ransfer to Disable . all Waiting to Disable .
Services Applications I Control Netw	Enable Hold Hold Format	Disable Send Only
■ GIP Definition ■ Coders and P ■ B GW and IP to	Held Timeout Call Hold Reminder Ring Timeout	-1 30
⊕	Enable Transfer Transfer Prefix	Disable 🗸
■ Routing ■ DTMF and DTMF & D	Enable Call Forward Enable Call Waiting	Disable V Disable V

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

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

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

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



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

 Message Waiting Indication (MWI) Parameter 	rs	
Enable MWI	Enable	\checkmark
MWI Analog Lamp	Disable	\checkmark
MWI Display	Enable	\checkmark
Subscribe to MWI	Yes	\checkmark
MWI Server IP Address	10.40.1.43:10060	
MWI Server Transport Type	UDP	\checkmark
MWI Subscribe Expiration Time	180	
Stutter Tone Duration	2000	
MWI Subscribe Retry Time	90	

Figure 25: Configure MWI parameters

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

•	✓ Conference									
4	Enable 3-Way Conference	Disable 🗸	[
	Establish Conference Code	!								
	Conference ID	conf								

Figure 26: Disable the 3-Way conference parameter



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

● Basic ● Full			_			
Gateway Port	U	ser Name		Passv	vord	
Port 1 FXS	167			5656	×	
Port 2 FXS						
 SIP Definitions Coders and Profiles GW and IP to IP Hunt Group Manipulations Routing DTMF and Supplementary Analog Gateway Keypad Features Metering Tones FXO Settings Authentication Automatic Dialing 	• Set • For (This is Note :	User Name to 167 . the password, enter the extension and p If you save and the Maintenance > So due to AudioCode will be erased. You here, after you res	5656 . assword en restore ftware U s' securit i will nee tore the	you set up on e the .ini file u Ipdate > Confij y concern, the ed to re-enter t .ini file.)	page 13.) nder guration File , password he password	

Figure 27: Configure Port 1 FXS gateway authentication

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

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

Gateway Port	User Name	Password
Port1 FXS	200xatgnemx5	*****
Port 2 FXS		
Port 3 FXS		
Port4 FXS		
Port 5 FXO		
Port 6 FXO		

(extension number)xatgnemx(E911 LID)

Figure 28: Configure e911 Location ID

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

200xatgnemx5

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



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

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

⊖ Basic ● Full		
€ Dystem		
DimovoIP		
⊕ @ Network		
Gecurity		
Gervices		
Applications Enabling		
🗉 🦾 Control Network		
	Register	Un-Register
Coders and Profiles	Sub	mit
GW and IP to IP		
B Hunt Group		
Endpoint Phone Number		
Hunt Group Settings		

Figure 29: Click Register

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

🝆 MaxAdministrator [AW67U1-]										
Services	System	PBX	Call Center	VoIP	Report	Diagnostic				
Login	Log	Jout	System	-	Trunk	Extension				
📦 Extension View										
					Rese	:t				
Extensi	on	Na	me		Rese Loca	tion				
Extensi	on	Na A1	IME		Rese Loca 03:00	tion 102(10.40.0.				

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

- 17. Attach an analog phone to the first FXS port and make a few calls to test that everything works correctly.
- 18. Submit this last change. On the toolbar, click **Burn**.
- 19. Restart the AudioCodes gateway.



FXO Configuration

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

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

Your environment must meet the following requirements:

- Your system must be running MAXCS Release 7.5.0.502 or later on an enterprise (on-premise) deployment. These configurations are not supported on a MAXCS Private Cloud deployment.
- No NAT is allowed between MAXCS and the gateway. For example, in Figure 1, IP addresses 10.40.1.43 (MAXCS) and the gateway (192.168.1.20) must be routable bi-directionally.
- MAXCS and the gateway can be on the same network.
- 1. In the AudioCodes configuration tool, click **Configuration** in the top left corner. Make sure that **Full** is selected for the menus.
- Select VoIP > GW and IP to IP > Hunt Group > End Point Phone Number. In addition to any extension numbers that may have already been configured, add trunk numbers. Note that empty rows between the FXS and FXO channels do not matter.

Basic • Full System System VoIP Media Gecurity Media Gervices Gervices Control Network		 Set entry 5 to channel 5. (On the MP-118, this is the first FXO port.) For channel 5, set the <i>Phone Number</i> field to your PSTN phone number (in our example, this is 4082520001) and set <i>Hunt Group ID</i> to 2 (FXO group). * Add more channels as needed; in this example, we have also configured channel 6. 					this is the first to your PSTN 2520001) and nple, we have			
⊕	Er	ndpoi	nt Phone Number	Table						
± Coders and Prof										
□@GW and IP to IP	1		Channe	l(s)	T	Phone Number	I	Hunt Group ID	T	Tel Profile ID
🗉 🖾 Hunt Group		1	1			167		1		0
Endpoint Pho		2								
Hunt Group :		3			Τ		Γ		Τ	
		4			Τ		Ι		Τ	
-	\rightarrow	5	5		Τ	4082520001	T	2	Т	0
		6	6		T	4082520002	T	2	T	0
		7			T		T		T	
		8			T		Ι			

Figure 31: Add trunk numbers to the Endpoint parameters

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



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



Figure 32: Configure Hunt Group 2

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



Figure 33: Configure source number to IP number manipulations



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

Basic • Full System VoIP CoIP Coint Security Coint Security Coint Security Coint Security Coint Security Coint Co	IP To Hunt Group Routin	Set Dest. Host Prefi and Source Phone P Set Source IP Addre Set Hunt Group ID t	x, Source Host Prefi. Prefix all to asterisk rss to the MAXCS IP ro 2 .	x, Dest. Phone Pref (*). address, 10.40.1.4	fix, 13.	
Applications Enabling Control Network SIP Definitions Coders and Profiles GW and IP to IP	Routing Index	ng Mode	1-12 V Route calls before ma	nipulation 🗸		Basic Pa
Hunt Group	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID
Routing	*	[167]	*	10.40.1.43		1
General Parameters						
Tel to IP Routing	*	*	*	10.40.1.43		2
IP to Hunt Group Ro						
Alternative Routing R	leasons					

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

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

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



Basic 🖲 Full		
System		
VoIP		
🗉 💷 Network		
B Cecurity		
B@@Media		
🗉 💭 Services		
Mapplications Enabling		
E Control Network	▼	
SIP Definitions	Dialing Mode	One Stage
Coders and Profiles	Waiting for Dial Tone	No
GW and IP to IP	Time to Wait before Dialing [msec]	1000
Hunt Group	Ring Detection Timeout [sec]	8
[™]	Reorder Tone Duration [sec]	255
Couting DTME and Supplement	Answer Supervision	No
□ Analog Gateway	Rings before Detecting Caller ID	1
Keypad Features	Send Metering Message to IP	No
Metering Tones	Disconnect Call on Busy Tone Detection (CAS)	Enable T
FXO Settings	Disconnect On Dial Tone	Enable 🔹
Authentication		

Figure 35: Configure FXO parameters

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

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



VoIP				
• Network				
E Security				
The Commission				
The Applications Each	line			
The Control Network	aing			
Control Network				
± SID Definitions				
E SIP Definitions	Automatic Dialing			
SIP Definitions Coders and Pr GW and IP to 1	Automatic Dialing			
SIP Definitions Coders and Pro GW and IP to I Hunt Group	Automatic Dialing			Hotline
SIP Definitions Coders and Pro GW and IP to I Hunt Group Maninulations	Automatic Dialing Gateway	Destination Phone	Auto Dial	Hotline Dial Tor
SIP Definitions Coders and Pre GW and IP to I Hunt Group Manipulations	Automatic Dialing Gateway Port	Destination Phone Number	Auto Dial Status	Hotlind Dial Tor Duratio [sec]
SIP Definitions Coders and Pro GW and IP to I GW Hunt Group Manipulations Routing DTME and Su	Automatic Dialing Gateway Port Port 1 FXS	Destination Phone Number	Auto Dial Status Enable V	Hotline Dial Tor Duratio [sec]
SIP Definitions Coders and Pro GW and IP to I GW Hunt Group Manipulations Routing DTMF and Su Analog Gates	Automatic Dialing Gateway Port Port 1 FXS Port 2 FXS	Destination Phone Number	Auto Dial Status Enable V	Hotlin Dial To Duratic [sec] 0
SIP Definitions Coders and Pro GW and IP to I GW Annipulations Routing DTMF and Su Analog Gatev Keypad Fea	Automatic Dialing Gateway Port Port 1 FXS Port 2 FXS Port 3 FXS	Destination Phone Number	Auto Dial Status Enable V Enable V	Hotlin Dial To Duratic [sec] 0 0
SIP Definitions Coders and Pro GW and IP to I Hunt Group Manipulations Routing DTMF and Su Analog Gates Keypad Fea Metering To	Automatic Dialing Gateway Port Port 1 FXS Port 2 FXS Port 3 FXS Port 4 FXS	Destination Phone Number	Auto Dial Status Enable V Enable V Enable V	Hotling Dial Tor Duratic [sec] 0 0 0
SIP Definitions Coders and Pro GW and IP to 1 Hunt Group Manipulations Coders and Pro Manipulations Coders Analog Gates Keypad Fea Metering To FXO Setting	Automatic Dialing Gateway Port Port 1 FXS Port 2 FXS Port 3 FXS Port 4 FXS Port 4 FXS	Destination Phone Number	Auto Dial Status Enable V Enable V Enable V Enable V	Hotlino Dial Tor Duratic [sec] 0 0 0 0

Figure 36: Configure automatic dialing rules for port 5

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

- 8. Submit your last changes. On the toolbar, click **Burn** to save this configuration.
- 9. Next, configure a SIP Group with SIP servers. Log into MAXCS Administrator and double-click **SIPSP** in *Boards* view. Click **Board Configuration** and then click **SIP Group Configuration**.
 - **Note:** If you are setting up multiple MP-1xx gateways for FXO lines, then you must create one Group for each gateway. The trunks may all be included in the same Out Call Route or have the same Trunk Access Code. You must also assign the number of SIP trunks to that group to match the number of FXO lines on the gateway.



	SIP Signaling Channel Configuration
Boards Image: Configuration 0 HMCP 1 MobileExtSP 2 SIPSP Board Logical ID	SIP Extension Channels 60 Current Configured Channels 60 Change Number of SIP Extension Channels to 60
Channel Mapping List Cogical Type Physical Channel Mapping List Cogical Type O SIP Extension SIP Extens	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) 16 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 Cancel

Figure 37: Open the SIP Group Configuration panel

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

IP Group Configuration					
Groups:			Add SIP Gr	oup	
N Fax Enabled AltiGen Trunk	Register Settings SIP OPT	IONS	General -		ОК
	Domain:			AltiGen Trunk	Cancel
	SIP Server IP Address:		-Fax Trun	k Routing	
	User Name:		Enat	ble fax trunk routing	
	Password:		Use	r Name:	
	Register Period:		Pass	sword:	
	SIP Source Port (Non-TLS):	<u> </u>		,	
	SIP Destination Port:				
Add Delete Edit					

Figure 38: Enter a name for this new SIP Group

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



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

IP Group Configuration	<	
Groups:		
Name Fax Enabled		
mysip No		
Add Delete Edit	Add SIP Server	
5IP Servers:	General	ОК
Domain Status F	Domain:	Capcel
		Cancer
perceptus up	Copy From	
	Group: N/A	
Add Del Up Down		
Refresh	Server: IN/A	

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

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

Register Settings SIP OPT	ions
Domain:	192.168.1.55
SIP Server IP Address:	192.168.1.55
User Name:	audiocodes
Password:	
Register Period:	0
SIP Source Port (Non-TLS):	10060 💌
SIP Destination Port:	10060

Figure 40: Configure SIP Trunk parameters



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

IP Ser	ver						
P Add	ress:	10.0.1	205				
IP Cal	ing Number			1			
SIP Pr	otocol Field:	FROM	1 Heade	H Contraction	-		
Custor	n P-Asserted-II		10000.00				
0) /				_		
Custon	n Liversion;						
C Ca C Ca Ca	inter can only a inter can only a illing Number c	ccept Callin ccept assign an be accep	g numb ned nur ted by	er with minin nbers as Cai Carrier	num 0 ling Numbe	_y dig er	its
с са с са с а	inier can only a inier can only a illing Number c From	ccept Callin ccept assign an be accep To	g numb ned nur ted by	er with minin nbers as Cal Carrier	num 0 ling Numbr	<mark>, y dg</mark> er Add	ts
60 60 60	mier can only a mier can only a illing Number c	ccept Callin ccept assign an be accep To	g numb ned nur ted by	er with minin nbers as Cai Carrier	num 0 ling Numb	→ dig er Add Edt	ts
C G G G G	mer can only a mer can only a lling Number c	ccept Callin ccept assign an be accep To	g numb hed nur ted by	er with minin nbers as Cai Carrier	num 0 ling Numb	r dg Add Edit Delete	its
C Ca C Ca D Ca	inter can only a inter can only a illing Number c From set this as Callin annot accept c	ccept Callin ccept assign an be accep To To ng Number if onfigured nu	g numb hed nur ted by the Ca mbers	er with minin hers as Cai Carrier	num 0 ling Numb	+ dig Add Edi Delete	ks
	mer can only a mer can only a lling Number c From From e this as Califr annot accept c and Caller Name	ccept Callin ccept assign an be accep To To ng Number if onfigured nu	g numb hed nur ted by the Ca mbers	er with minin hbers as Cal Carrier nier Enable Sta	num 0 ling Numb-	Add Edit Delete	its
	mer can only a mer can only a illing Number c From From e this as Califr annot accept c and Caller Name able SIP REFE	ccept Callin ccept assign an be accep To To ng Number if onfigured nu e	g numb ned nur ted by the Ca mbers	er with minin hbers as Cal Carrier mer Enable Sta Enable Ce	num 0 ling Numb-	Add Edit Delete	its
C Ca Ca Ua F Se F En Incomin	mer can only a mer can only a ling Number c From From set his as Calin annot accept c and Calier Name able SIP REFE ig DID Number	ccept Callin ccept assign an be accept To To g Number if onfigured nu s ER	g numb ned nur ted by the Ca mbers	er with minin nbers as Cal Carrier I tier Enable Sta Ervable Ce	num 0 ling Numb	dig d	its

Figure 41: Change the SIP Protocol Field

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

Enabled	ID	Channel No	Group	~	Max SIP Trunk License
2	36	148	Audiocodes		62
1	37	149	Audiocodes		
2	38	150	Audiocodes		Assigned SIP Trunk License
2	39	151	Audiocodes		38
2	40	152	Audiocodes		
2	41	153	Audiocodes		Assian Group
2	42	154	Audiocodes		
2	43	155	Audiocodes		If you need more channels, close this
2	- 44	156	Audiocodes		panel, click the 'Board Configuration'
2	45	157	Audocodes		button, and set an appropriate value for "Chappe the Member of SIP To of
2	46	158	Audiocodes		Channels to". Then restart the
2	47	159	Audiocodes		system.
2	48	160	Audocodes		
	49	161		1	
	50	162			
	51	163			
	52	164			
	53	165			
	54	166			
	55	167			
	56	168			
	57	169			
	58	170		4	
<		101		>	OK Cancel

Figure 42: Enable channels

- 15. Now you can add these newly-created trunks to an outbound route or assign them to a trunk access code.
- 16. At this point, test that you can make outbound calls and inbound calls from MAXCS using your IP phone and an FXS port. Do not continue to the next section until these calls are working correctly.



Fax-over-IP Configuration (Optional)

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

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

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

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

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

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

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

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

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

SIP Gener	ral Parameters	
-	SIP General	
4	NAT IP Address	0.0.0.0
	PRACK Mode	Supported
	Channel Select Mode	By Dest Phone Number 🔹
	Enable Early Media	Disable 🔹
	Session-Expires Time	0
	Minimum Session-Expires	90
	Session Expires Method	re-INVITE 🔹
	Asserted Identity Mode	Disabled 🔹
	Fax Signaling Method	T.38 Relay 🔹
	SIP Transport Type	UDP 🔹
	SIP UDP Local Port	10060

Figure 43: Configure SIP general parameters



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

Fax/I	Nodem/CID Settings	
	Fax Transport Mode	T.38 Relay
	Caller ID Transport Type	Mute 🔻
	Caller ID Type	Standard Bellcore
	V.21 Modem Transport Type	Disable T
	V.22 Modem Transport Type	Enable Bypass 🔻
	V.23 Modem Transport Type	Enable Bypass 🔻
	V.32 Modem Transport Type	Enable Bypass 🔻
	V.34 Modem Transport Type	Enable Bypass 🔻
	Fax CNG Mode	Sends on CNG or v8-cn 🔹
	CNG Detector Mode	Relay
	See Deless Collins	-
	✓ Fax Relay Settings	0
	Fax Relay Redundancy Depth	U
	Fax Relay Enhanced Redundancy Depth	4
	Fax Relay ECM Enable	Enable V
	Fax Relay Max Rate (bps)	14400bps T
	- Buppes Settings	
	Dypass Settings	0711114/mm
	Fax/Modem Bypass Coder Type	Grinnuaw
	Fax/Modem Bypass Packing Factor	1
	Fax Bypass Output Gain	0
	Modem Bypass Output Gain	0

Figure 44: Set Fax parameters

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

PSTN Survival (SAS) Configuration (Optional)

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

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



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

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

PSTN SAS General Configuration

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

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

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

🔿 Basic 💿 Full	0		
±@System	Applications Enabling		
DIP			
	•		
	🗲 SAS Application	Disable 🔻	
. ⊕ 💭 Media		Disable	
• 🖾 Services			
Califications En	abling	· · · · · ·	
Applications Er	nabling		

Figure 45: Enable the SAS application

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

Follow these steps after the device has restarted:

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



◯ Basic	Proxy Sets Table					
⊕	→	▼ Proxy Set ID		9		~
Network Security Media Services Applications Enabling Control Network IP Group Table Proxy Sets Table		1 192.168 2 3 4 5	Proxy Addre:	55	Transport Type	
		▼				
		Enable Proxy Keep A	Alive	Disable		~
		Proxy Keep Alive Tir	ne	60		
		Proxy Load Balancin	g Method	Disable		~
		Is Proxy Hot Swap		No		~
		Proxy Redundancy N	1ode	Not Configure	ed	~

Figure 46: Configure the Proxy Sets table

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

□ 🖾 VoIP			
± Security	•		
Decunty	SAS Local SIP UDP Port	5060	
± 💷 Media	SAS Default Gateway IP		
Gervices	SAS Registration Time	90	
- Applications Enabling	SAS Local SIP TCP Port	5060	
	SAS Local SIP TLS Port	5081	
Control Network	SAS Proxy Set	9	
	SAS Emergency Numbers		
T Coders and Profiles	SAS Binding Mode	1-User Part Only	~
	SAS Survivability Mode	Auto-answer REGISTER	~
GW and IP to IP	Enable ENUM	Disable	~
Bas	Enable Record-Route	Disable	~
Stand Alone Survivability	SAS Block Unregistered Users	Un-Block	~
	Redundant SAS Proxy Set	-1	
	SAS Connection Reuse	Enable	~
	SAS Inbound Manipulation Mode	None	~

Figure 47: Set SAS parameters

• Set SAS Local SIP UDP Port to 5060



- Set SAS Registration Time to 90
- Set SAS Proxy Set to 9
- Set SAS Binding Mode to 1-User Part Only
- Set SAS Survivability Mode to Auto-answer REGISTER
- 5. Select VoIP > GW and IP to IP > Manipulations > Dest Number IP -> Tel. You will add three new rules.
- 6. Click **Add** to add the first rule. Set the parameters on the *Rule* tab as shown in the next figure. There are no parameters to set on the *Action* tab.

Rule Action	
Index	0
Destination Prefix	*
Source Prefix	*
Source IP Address	10.40.1.43
Source Host Prefix	*
Destination Host Prefix	*

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

Figure 48: Set destination number manipulation rule 1

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

Rule Action	
Index	1
Destination Prefix	911
Source Prefix	*
Source IP Address	*
Source Host Prefix	*
Destination Host Prefix	*

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

Figure 49: Set destination number manipulation rule 2

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



Rule Action	
Index	Þ
Destination Prefix	9
Source Prefix	*
Source IP Address	*
Source Host Prefix	*
Destination Host Prefix	*

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

Rule Action	
Index	2
Stripped Digits From Left	1
Stripped Digits From Right	0
Number of Digits to Leave	255
Prefix to Add	
Suffix to Add	
Presentation	Not Configured 🗸

Figure 50: Set destination number manipulation rule 3

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

O Basic ⊕@Sys ⊖@Vol	• sten	Full							
	IF	To Hunt Group Routing 1	Table					1	
			Routing Index IP To Tel Routin	g Mode	1-12 ▼ Route calls before ma	nipulation 🗸			
		Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID	IP Profile
	1	•		[167]		! •		1	0
	2	•		ŀ		·		2	0
	3				ļ				0
	G Te IP Al	eneral Parameters el to IP Routing to Hunt Group Routir ternative Routing Rea	ng isons			14	_		

Figure 51: Configure IP to Hunt Group routing rules

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



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

- IMPORTANT!Because the wildcard character is being used, be aware that any SIP device routable to
the device could use the device to place PSTN toll calls. To prevent fraud calls, make
sure your MP-11x device is behind a firewall. Also, no SIP port forwarding to the MP-
11x device should be configured on the firewall, or else malicious users from the in-
ternet can use the device to place PSTN calls.
- b) For other calls, route the call to FXO ports (Huntgroup ID 2). If there is an existing entry for this, change the *Source IP Address* to an asterisk (*). if there is no such entry, add an entry 2 as follows:
 - Set Destination Host Prefix, Source Host Prefix, Destination Phone Prefix, Source Phone Prefix, and Source IP Address to an asterisk (*).
 - Set Hunt Group ID to 2.

At this point, the general configuration is complete.

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

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

SAS LAN Deployment Architecture

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

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





Figure 52: Architecture for LAN environment

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

SAS FXO Configuration for Cloud Deployments

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

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

Cloud Deployment Architecture

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







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

Requirements

Your environment must meet the following requirements:

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

Prerequisites

Perform these steps before you proceed:

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



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

Configuration Procedures

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

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

Basic Full System		
Volp	Common Gateway	
	Index	1
Applications Enablir	Description ->	LocalFXO
IP Group Table	Proxy Set ID ->	1
Proxy Sets Table	SIP Group Name	
	Contact User	
	Local Host Name	
	Media Realm Name	
	IP Profile ID	0

Figure 54: IP Group table parameters

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



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

p 🚔 👝 🔒	Proxy Sets Table				
" System	-				
□//// VoIP	Proxy Se	t ID	1		~
Network Network Security Media Services Applications Enabling Control Network	->	Proxy Addres 1 192.168.1.20:5060 2 3 4 5 5		Transport Type	
IP Group Table					
Proxy Sets Table	-				
	Enable Pr	roxy Keep Alive	Disable		~
	Proxy Ke	ep Alive Time	60		
	Proxy Loa	ad Balancing Method	Disable		~
	Is Proxy	Hot Swap	No		~
	Proxy Re	dundancy Mode	Not Configure	d	~

Figure 55: Proxy Sets table parameters

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

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

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

⊖ Basic ● Full		\bigcirc			
±@ System	Endpoi	nt Phone Number Table			
□@VoIP					
• Network		Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
	1	1	167	1	0
🗉 💷 Media	2				
	3				ii ii
E Application	4				T T
Control Net	5	5	4082520001	2	0
SIP Definiti Definiti	6	6	4082520002	2	0
GW and IP	7				
Hunt Group Endpoint Hunt Grou) Phone N Ip Settii	lumber			

Figure 56: Endpoint Phone Number table



Set the following parameters:

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

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

- 6. Add more channels as needed; in Figure 56, channel 6 has also been configured.
- 7. Select VoIP > GW and IP to IP > Hunt Group > Hunt Group Settings. Configure a hunt group ID 2 as follows.



Figure 57: Hunt Group 2

- Set entry 2 to Hunt Group ID 2.
- Set Channel Select Mode to Cyclic Ascending.
- Set *Registration Mode* to **Don't Register**.
- Set Serving IP Group ID to **1**.
- 8. Select VoIP > GW and IP to IP > Analog Gateway > FXO Setting. Set *Dialing Mode* to One Stage; set *Disconnect on Dial Tone* to Enable.



🗆 Basic 🔍 Full			
€ System			
□ 🖾 VoIP			
⊞ @_Media			
■ Services			
Applications Enabling			
Control Network	▼		
■■SIP Definitions	Dialing Mode	One Stage	•
Coders and Profiles	Waiting for Dial Tone	No	•
GW and IP to IP	Time to Wait before Dialing [msec]	1000	
Hunt Group	Ring Detection Timeout [sec]	8	
Manipulations	Reorder Tone Duration [sec]	255	
DTMF and Supplement	Answer Supervision	No	T
🖃 👜 Analog Gateway	Rings before Detecting Caller ID	1	•
Keypad Features	Send Metering Message to IP	No	•
Metering Tones	Disconnect Call on Busy Tone Detection (CAS)	Enable	۲
FXO Settings	Disconnect On Dial Tone	Enable	•
Authentication			_

Figure 58: Configure FXO parameters

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

SAS IP Phone Configuration

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

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

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

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

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

Prerequisites

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

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



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

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

🕘 Basic 🖲 Full	\odot		
± @ System			
DIVOIP			
• Detwork			
• Decurity			
• Media			
⊕ @ Services			
Applications Enabling			
E Control Network	Automatic Dialing		
■ Coders and Profiles			_
GW and IP to IP	Gateway	Destination Phone	
🗉 💷 Hunt Group	For	Number	
Manipulations	Port 1 FXS		
# Routing	Port 2 FXS		
DTMF and Supplementary	Port 3 FXS		
□@@Analog Gateway			
	Port 4 EXS		
Keypad Features	Port 4 FXS	167	
Keypad Features Metering Tones	Port 4 FXS Port 5 FXO	167	
Keypad Features Metering Tones FXO Settings	Port 4 FXS Port 5 FXO Port 6 FXO	167 167 167	
Keypad Features Metering Tones FXO Settings Authentication	Port 4 FXS Port 5 FXO Port 6 FXO	167 167 167	
Keypad Features Metering Tones FXO Settings Authentication Automatic Dialing	Port 4 FXS Port 5 FXO Port 6 FXO	167 167	

Figure 59: Set Automatic dialing parameters

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

🔺 A L T I G E N

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



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

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

Secondary Proxy Enable Secondary Proxy			
Address:	192 . 168 . 1 . 20		
Port:	5060		
Enable Registration			

Figure 60: Polycom Secondary Proxy settings

- 4. For the MP-11x FXS port acting as an IP extension:
 - a) In the gateway web configuration tool, select VoIP > Control Network > Proxy Sets Table.
 - b) Configure Proxy Set ID **0**:
 - a. *Proxy Address* entry 1 should already have been configured, because your extension on the FXS port was already working while MaxCS is reachable.
 - b. For Proxy Address entry 2, enter 192.168.1.20:5060.
 - c. Set *Enable Proxy Keep Alive* to **Using Options**.
 - d. Set Proxy Keep Alive Time to 15.
 - e. Set Proxy Redundancy Mode to Homing.





Figure 61: Proxy Set ID 0 parameters

- c) Submit the change and click **Burn**.
- d) You must modify the registry. On the MaxCS server, run *regedit* as an administrator.
- e) Export your current registry to a file, as a backup. Be sure to follow Microsoft's recommended procedures for updating your registry.

Name	Туре	Data
赴 (Default)	REG_SZ	(value not set)
🕮 BlockSIPUnauthorizedCall	REG_DWORD	0×00000000 (0)
ab) clsid	REG_SZ	{DB22B6B0-32CB-11d5-8A10-0050DA71971
ab ConfigDllPath	REG_SZ	C:\AltiServ\SP\SIPSP\SIPConfigDialog.dll
ab DilPath	REG_SZ	C:\AltiServ\SP\SIPSP\SIPSP.dll
EnableSIPOptionBehindNAT	REG_DWORD	0×00000001 (1)
🕮 MaxBackFiles	REG_DWORD	0x00000032 (50)
🕫 SipDIDExt	REG DWORD	0×00000001 (1)

Figure 62: Update the registry

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

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

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

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



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

- g) Save these changes.
- h) Restart the MaxCS services (or reboot the MaxCS server).

Confirm the Configuration

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

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

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

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

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

- 3. From the phone, dial 915102520001 and press #. The call should connect properly.
- If you enabled registration for the phone, log into the MP-11x web configuration tool and click Status and Diagnostics above the menus. Then select VoIP Status > SAS/SBC Registered Users. You should see extension 167 in the Address of Record field.

SA	S/SBC Registered Users	
	Address Of Record	Contact
	167	< 167 @ OPTIONS, II REFER"

Figure 63: Registered users shows extension 167

- 5. Make a call to 14082522001 (this is the FXO port's PSTN number) from your mobile phone. IP phone 167 should ring. Pick up the call and verify the connection.
- 6. Restart the MaxCS server. IP phone 167 should reconnect to the MaxCS server shortly. The length of the delay varies greatly, depending upon the size of the system; on small systems this may take 2-3 minutes; on large MaxCS systems it may take up to 10 minutes.
- Configure another IP Phone with extension 168 with registration turned on. Shut down the MaxCS server again and check that the extension is registered (just as you did in step d. Make a call from IP phone 168 to IP phone 167; answer the call and verify the connection.



Centrex Configuration (Optional)

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

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

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

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

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

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

1. In your browser, navigate to this URL:

http://192.168.1.20/AdminPage

Substitute the gateway's IP address for 192.168.1.20.

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



Figure 64: Specify INI parameters

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



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

(Basic 🖲 Full	IPMedia :	Settings		
	⊕ 🖾 System ⊟ 🚾 VoIP				
	Network	÷	IPMedia Detectors	Enable	-]
	Elecurity		Enable Answer Detector	Disable 🗸	·
	🗆 💷 Media		Answer Detector Activity Delay	300]
	Voice Settings Fax/Modem/CID Se RTP/RTCP Settings IPMedia Settings General Media Sett		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 65: Adjust IP Media parameter

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

Basic Full	Supplementary Services		
Generations Services Control Network Generations Gener	 Enable Hold Hold Format Held Timeout Call Hold Reminder Ring Timeout Enable Transfer Transfer Prefix Enable Call Forward Enable Call Waiting 	Enable 0.0.0.0 -1 30 Enable Disable Disable	> > > >
DTMF and Supplementary	~		

Figure 66: Adjust Supplementary Services parameters

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

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



Figure 67: Create a new SIP Trunk Profile

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

SIP Trunk - Id=15, Logical	×	
SIP Server IP Address User Name Password	192.168.1.20 audiocodes	
Domain	192.168.1.20	
SIP Register Period	0	Sec.
SIP Trunk Profile 🛛 🗕 🗕	centrex 💌	
SIP Source Port (Non-TLS)	10060 💌	
SIP Destination Port	10060	
🔽 Enable Channel		

Figure 68: Set SIP Trunk Profile to centrex



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

Restriction	Answering	One Number Access	Monitor List
General	Group Speed	Dialing 🔰 Mail Managemen	t Notification
- Personal Informal	tion		
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 Inter	com 🗖 Agent 🎽 🔽 Release	e SIP Tie-Link Trunk

Figure 69: Configure extension 150

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

Other Call Restrictions
Allow Calls to be Transferred or Conferenced to an Outside Number
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 70: Set call restriction options for extension 150

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

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

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

General Restriction	Group Ans	Speed Dialing wering
Forward All Cal	is rward to Free Forr 5102520001,	nat

Figure 71: Set forwarding to Free Format



- 14. Follow these steps to verify that the configuration is correct:
 - a) From a mobile phone, make a call to the AudioCodes FXO port. In this example, the number is 14082520001.
 - b) Confirm that the call routes to the AltiGen's IVR system.
 - c) In the IVR system, dial the virtual extension number you just created extension 150.
 - d) PSTN phone 15102520001 should ring. Answer the call and confirm that you can hear voice.
 - e) The SIP trunk in MAXCS should be available.

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

Figure 72: Inject numbers after the IVR system answers

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

Troubleshooting Tips

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

TLS Configuration (Optional)

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

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

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



• You cannot configure both Centrex and TLS on the same MP-1xx device.

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

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

O Basic ● Full ⊕@_System	©		
□@VoIP	✓ General Media Security Settings		
■ ■ Network	🗲 Media Security	Enable	~
Security Andia	Media Security Behavior	Preferable	~
Voice Settings	Authentication On Transmitted RTP Packets	Active	~
Fax/Modem/C	Encryption On Transmitted RTP Packets	Active	~
RTP/RTCP Set	Encryption On Transmitted RTCP Packets	Active	~
IPMedia Settir	SRTP Tunneling Authentication for RTP	Disable	~
Analog Setting	SRTP Tunneling Authentication for RTCP	Disable	~
Media Realm T	able		
Media Security			
Media Ouality	of experience		

Figure 73: Enable media security

🔼 A L T I G E N

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

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



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



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

O Basic Full	<u>></u>					
± ∭ System	Proxy Sets Table					
□///VoIP		-				
• Metwork		Proxy Set I	D	0		~
		TTOXY OCCI	<i>.</i>	<u> </u>		-
⊕@_Media						
± Services			Proxy Add	ress	Transport Type	
		1	10.40.1.43:5061		TLS 🗸	
Castral Network		2				
Control Network						
IP Group Table		3				
Proxy Sets Table		4				
		5				
			L		J	

Figure 75: Edit proxy set 0

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

Continue with one of the next two sections as appropriate.

TLS Configuration for an FXS Port

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

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

- 3rd Party SIP Device
Enable SIP Telephony Service
Enable Polycom Advanced Features
SIP Transport
Persistent TLS 🔽 SRTP
J

Figure 76: Check the SIP Transport options

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



TLS Configuration for an FXO Port

Follow these steps to enable TLS for an FXO port.

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

Name:	mp1xxtls
	Selected Codec
	G.711 Mu-Law
Codec:	
DTMF Delivery	RFC 2833
SIP Early Media	Enable
	(

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

Figure 77: Configure the new codec profile

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

💦 Edit IP Device Range 🛛 🗙		
From:	192.168.1.20	
To:	192.168.1.20	
Codec:	MP11x TLS	
ОК	Cancel	

Figure 78: Select the codec profile

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



📴 Boards		STP Signaling Channel Configuration
Logic Board Type	Board Configuration	SIP Extension Channels
MobileExtSF	Board Info Board Logical ID	Current Configured Channels 60
1	Channel Mapping List Logical Type Physical Channel Grou U SIP Extension SIP Ext	SIP Tie-Trunk Channels (Connecting AltiServ-to-AltiServ VolP 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 12 SIP Group Configuration Channel Assignment Advanced Configuration Channel Assignment *Note: Changing number of SIP extension or tie trunk channels requires stop and re-start switching and gateway services. OK Cancel
Register Settings	SIP OPTIONS	
Domain: SIP Server IP Addre User Name: Password: Register Period: SIP Source Port (Na SIP Destination Por	mysip ess: audiocodes •••••• 0 on-TLS): 10060	

Figure 79: Update the SIP Destination port

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

Resetting the Gateway to Default Settings

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

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



Configuration Maintenance Status & Diagnostics	Authentication		
Scenarios Search	Gateway Port	User Name	Password
Basic • Full	Port 1 FXS	167	xxxxxx
Control Network	Port 2 FXS		
General Parameters	Port 3 FXS		
Advanced Parameters	Port 4 FXS		
Account Table	Port 5 FXO		
RADIUS Accounting Settings	Port 6 FXO		
Coders and Profiles	Port 7 FXO		
GW and IP to IP	Port 8 FXO		
Endpoint Phone Number			
Hunt Group Settings			
Manipulations			
Routing DTME and Supplementary			
China and Supplementary			
Keypad Features			
Metering Tones			
FXO Settings			

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

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

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

Figure 81: Save the current configuration

- 2. Reset the device to its default settings. This procedure will **not** reset the device's web login IP address.
 - a. Use the Windows application Notepad to create an empty file. In this example, we will name it *null-config.ini*.
 - In the AudioCodes configuration tool, click Maintenance above the menu, and then select Software Update > Configuration File. Click Choose File and select the empty file that you created in the previous step.
 - c. Click Load INI File and follow the instructions to reboot the gateway.



Known Limitations and Workarounds

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

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

For this reason, you should make sure that all network cables and uplink switches are connected **before** you turn on your gateway.

• On an MP-118 gateway with a Level 3 uplink switch (such as Dell), it can take up to a full minute before the switch provides the network connection to the gateway. This delay is due to certain spanning tree settings on the switch, and may cause some channels not to register during the boot sequence.

To avoid such problems, AltiGen recommends that you insert a regular switch between the Level 3 switch and the MP-118 gateway.

• If your network is not stable, the instability may cause some of the channels on the MP-124 to not register after a network outage.

AltiGen Technical Support

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

For general configuration information for your gateway device, refer to your AudioCodes documentation. To find your AudioCodes user manual, search for "LTRT-65417 MP-11x and MP-124 SIP User's Manual Ver 6.6.pdf" in your web browser.

AltiGen provides technical support to Authorized AltiGen Partners and distributors only. End user customers, please contact your Authorized AltiGen Partner for technical support.

Authorized AltiGen Partners and distributors may contact AltiGen technical support by the following methods:

- You may request technical support on AltiGen's Partner web site, at https://partner.altigen.com. Open a case on this site; a Technical Support representative will respond within one business day.
- Call 888-ALTIGEN, option 5, or 408-597-9000, option 5, and follow the prompts. Your call will be answered by one of AltiGen's Technical Support Representatives or routed to the Technical Support Message Center if no one is available to answer your call.

Technical support hours are 5:00 a.m. to 5:00 p.m., PT, Monday through Friday, except holidays.

If all representatives are busy, your call will be returned in the order it was received, within four hours under normal circumstances. Outside AltiGen business hours, only urgent calls will be returned on the same day (within one hour). Non-urgent calls will be returned on the next business day.

Please be ready to supply the following information:



- Partner ID
- AltiGen Certified Engineer ID
- Product serial number
- AltiWare or MAXCS version number
- Number and types of boards in the system
- Server model
- The telephone number where you can be reached