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MAXCS Release 7.0 Update 1

Configuring Analog Extensions with AudioCodes Gateways

Intended audience:

AltiGen Authorized Partners

May 7, 2014



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About This Guide

This guide explains how to configure one or more analog extensions using an AudioCodes VoIP SIP MP-11x or MP-124 gateway.

These instructions will work behind most on-premise firewalls, NAT routers, and EdgeMarc SBCs. MaxCS and the AudioCodes gateway can be on the same LAN or can be inter-connected via VPN.

The examples in this guide illustrate the configuration for an AudioCodes MP -118 gateway. Refer to the documentation for your gateway model as needed (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).

This guide comes with two files. (These two files can be found in the AltiGen Knowledge Base, along with this guide.) Store these files locally; you will need to upload them into the AudioCodes configuration tool during the procedures in this guide, in the section [Verification](#).

- usa_tones_13_NoHold.ini
- usa_tones_13_NoHold.dat

A support agreement with AudioCodes is required.

Note: AltiGen does not provide general configuration support for AudioCodes products.

Related Documents

- For instructions on configuring remote survivability, refer to *Application Note: Remote Survivability with AudioCodes Gateways*. This guide is available from your AltiGen representative.
- For instructions on configuring Polycom IP phones, refer to the *Polycom IP Phone Configuration Guide*.

Introduction

When configured, MaxCS treats analog extensions behind an AudioCodes gateway as 3rd-party IP phones. The analog phones connect to an FXS port. The FXS port converts the analog signal to SIP.

The next figure depicts the configuration followed in this guide:

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

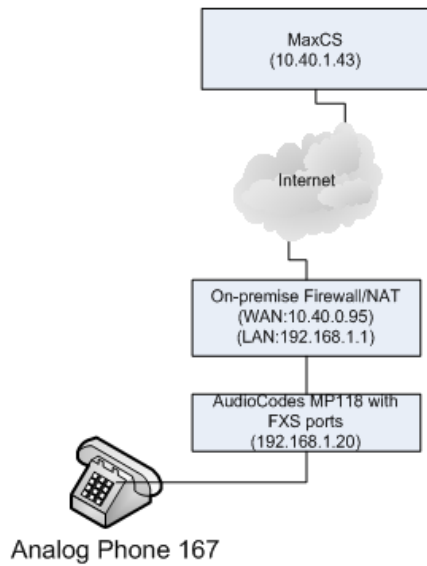


Figure 1: Diagram of MaxCS, Firewall/NAT, and AudioCodes gateway

Prerequisites

To set up analog extensions using an AudioCodes MP-11x or MP-124 gateway, you must meet the following requirements:

- You must be running either MaxCS Release 7.0 Enterprise or MaxCS Private Cloud.
- Proper configuration for MaxCS behind NAT is required (including the port forwarding and Enterprise Manager settings). Any AltiGen IP phones behind NAT should already be working correctly behind the on-premise firewall.
- The AudioCodes gateway must already have a static IP address and must be able to be configured through the web interface.
- Make sure that your AudioCodes gateway is running the correct version of firmware: Click the **Home** button and check that you have firmware version 6.60A.265.010 or later.
- You must have a valid MaxCS 3rd-party SIP license to implement analog support.

IMPORTANT! If you plan to have your MP-118 gateway support both FXS and FXO, we recommend that you to configure your analog extensions first, and then configure survival mode. A proxy group is required when you are running both FXS and FXO on the same gateway.

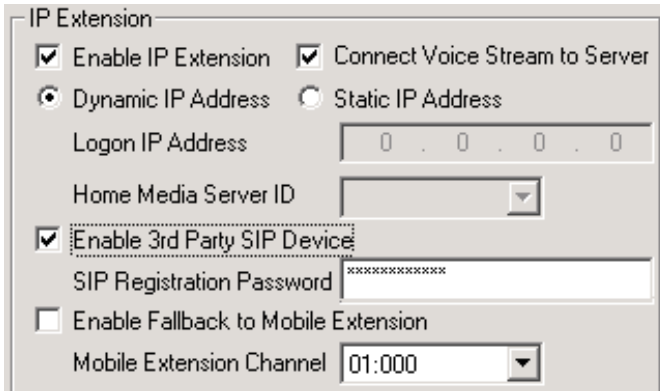
MaxCS Configuration

This section describes the configuration steps for MaxCS 7.0. Refer to Figure 1 as needed.

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

1. Log into MaxCS 7.0 MaxAdministrator, select **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 2: Configure IP Extension settings

3. Select **PBX > Altigen IP Phone Configuration**.

4. For extension 167, clear the checkbox **Enable SIP Telephony Service**.

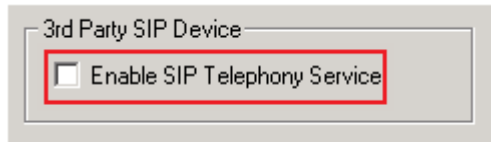
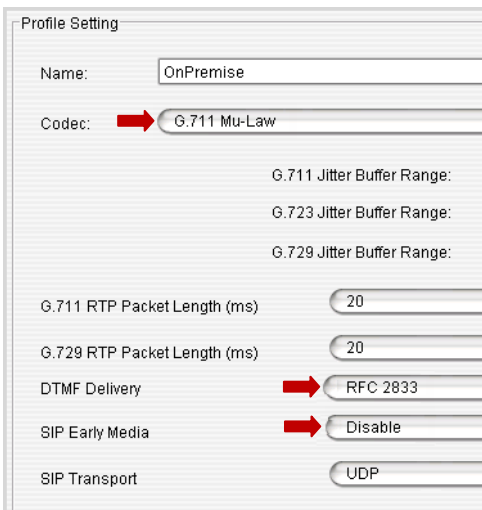


Figure 3: Disable SIP Telephony Service

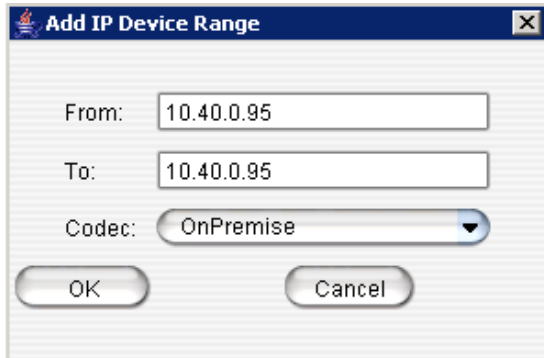
5. Select **VoIP > Enterprise Network Management > Codec > Codec Profile Table**. Add a new profile, "OnPremise."



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

Figure 4: Create a new codec profile

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



Add IP Device Range

From: 10.40.0.95

To: 10.40.0.95

Codec: OnPremise

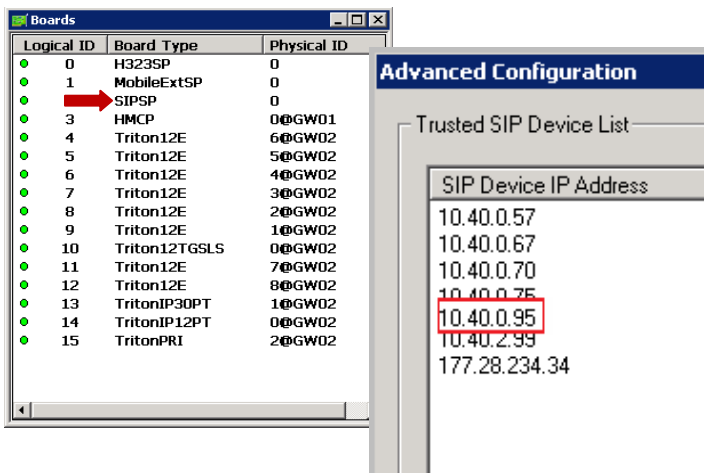
OK Cancel

- Set both the *From* and the *To* fields to **10.40.0.95** (as shown in Figure 1)
- For the *Codec*, select **OnPremise**.

Figure 5: Configure the IP Device range

7. Open the MaxAdministrator *Boards* view. Double click the **SIPSP** entry, select **Board Configuration**, and click **Advanced Configuration**. Add **10.40.0.95** to the *Trusted SIP Device List*.

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



Logical ID	Board Type	Physical ID
0	H323SP	0
1	MobileExtSP	0
2	SIPSP	0
3	HMCP	0@GW01
4	Triton12E	6@GW02
5	Triton12E	5@GW02
6	Triton12E	4@GW02
7	Triton12E	3@GW02
8	Triton12E	2@GW02
9	Triton12E	1@GW02
10	Triton12TGSLs	0@GW02
11	Triton12E	7@GW02
12	Triton12E	8@GW02
13	TritonIP30PT	1@GW02
14	TritonIP12PT	0@GW02
15	TritonPRI	2@GW02

Advanced Configuration

Trusted SIP Device List

SIP Device IP Address
10.40.0.57
10.40.0.67
10.40.0.70
10.40.0.75
10.40.0.95
10.40.2.99
177.28.234.34

Figure 6: Add the firewall/NAT public address to the Trusted SIP Device list

AudioCodes MP-118 Configuration

Important! While working in the AudioCodes configuration application, make sure that **Full** is always selected for the *Configuration* menu, so that you can see all of the menu options. Check this setting after you log out and log back in, as sometimes the application can reset the menus to **Basic** mode.

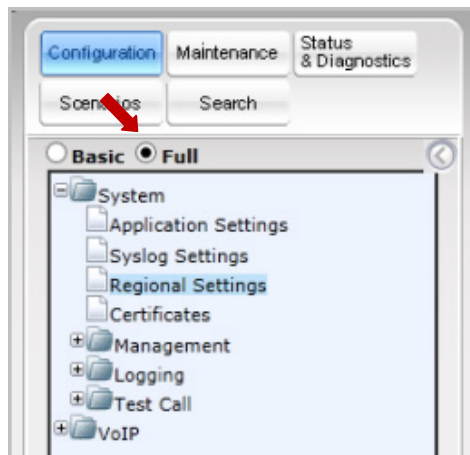


Figure 7: Choose Full for the AudioCodes Configuration menu

Also, make sure you click **Submit** after each change, and later click **Burn**, or your configuration changes will be lost after the gateway is restarted.

1. Click **Configuration** in the top left corner (see Figure 7). 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**.)

2. Select **VoIP > Network > IP Routing Table** and confirm that the IP address is 192.168.1.20 (the AudioCodes IP address from Figure 1), the prefix length, and the network's gateway IP addresses are set up properly. (This is a read-only page.)

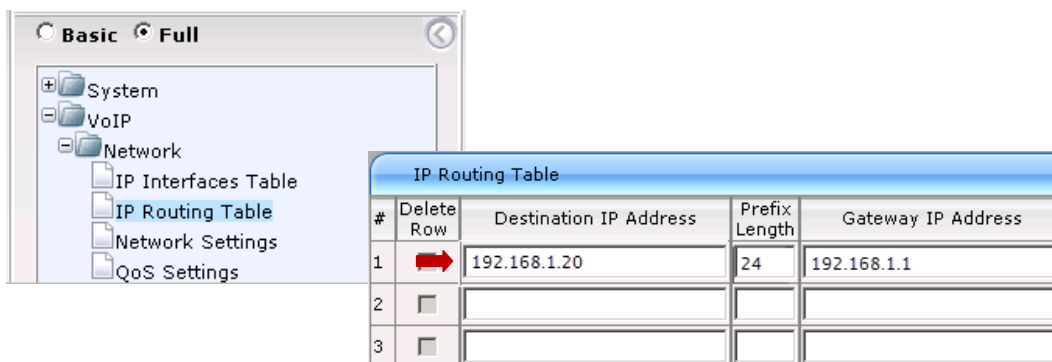
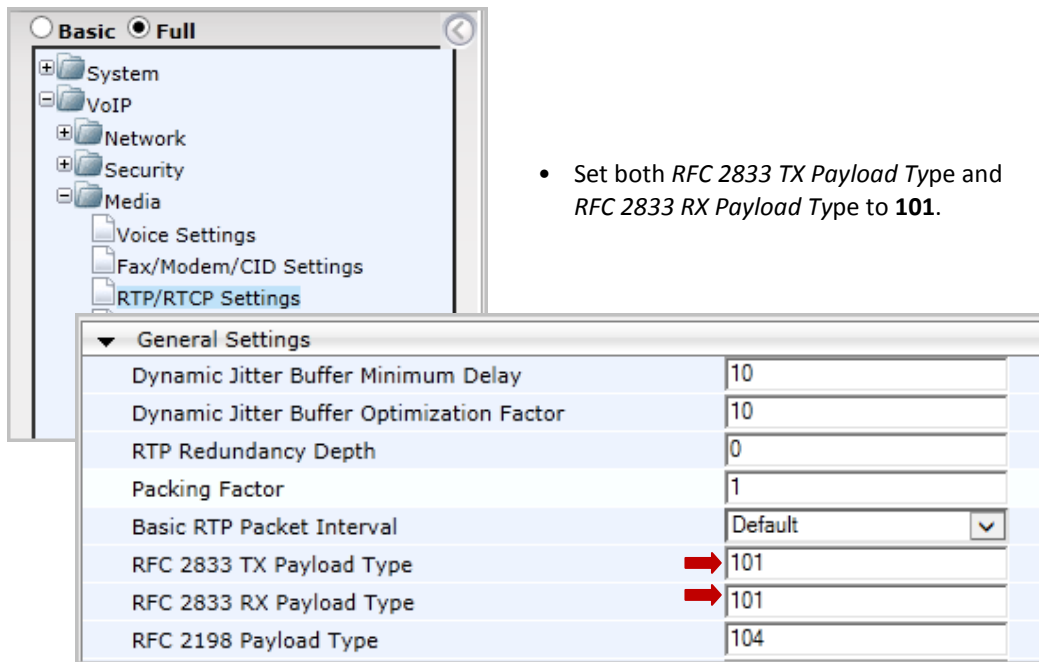


Figure 8: Check the Network IP Interface settings



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



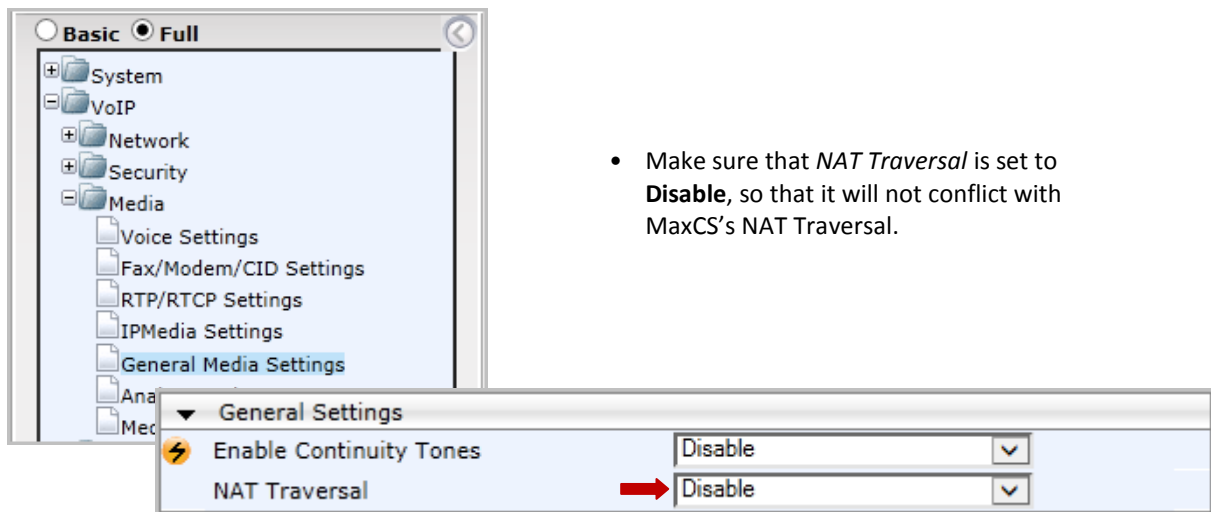
The screenshot shows the configuration window for RTP/RTCP Settings. The left sidebar has a tree view with 'Media' expanded and 'RTP/RTCP Settings' selected. The main area shows 'General Settings' with the following values:

Setting	Value
Dynamic Jitter Buffer Minimum Delay	10
Dynamic Jitter Buffer Optimization Factor	10
RTP Redundancy Depth	0
Packing Factor	1
Basic RTP Packet Interval	Default
RFC 2833 TX Payload Type	101
RFC 2833 RX Payload Type	101
RFC 2198 Payload Type	104

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

Figure 9: Configure the Media RFC 2833 Payload settings

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



The screenshot shows the configuration window for General Media Settings. The left sidebar has a tree view with 'Media' expanded and 'General Media Settings' selected. The main area shows 'General Settings' with the following values:

Setting	Value
Enable Continuity Tones	Disable
NAT Traversal	Disable

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

Figure 10: Configure the general media settings

5. 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]	300
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 11: Configure the minimum and maximum detection periods

Note: The AudioCodes device requires a reboot after you change either of these settings. You can reboot once all configuration is complete, during the last step on page 17.

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

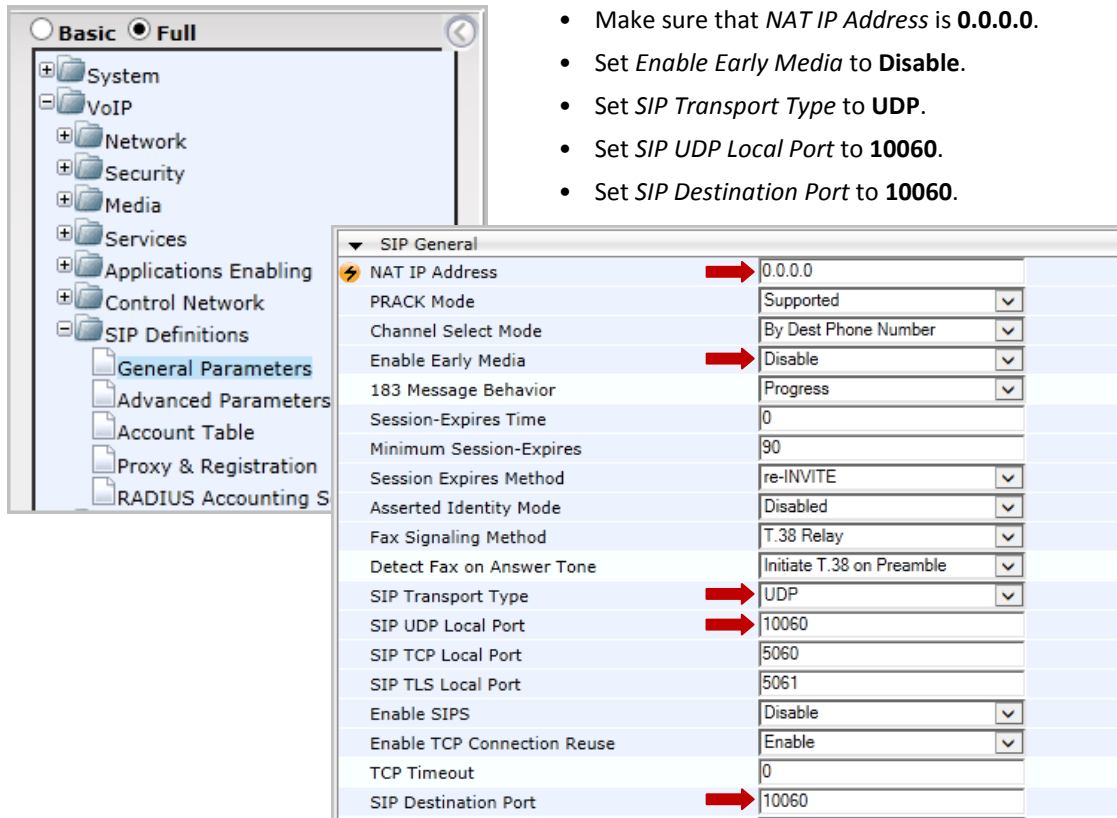
○ Basic ● Full	
+	System
-	VoIP
+	Network
+	Security
+	Media
+	Services
+	Applications Enabling
-	Control Network
	IP Group Table
	Proxy Sets Table

- Set *Proxy Address 1* to **10.40.1.43:10060** (The MaxCS public IP address and port). Set its *Transport Type* to **UDP**.
- Set *Enable Proxy Keep Alive* to **Disable**.

▼ Proxy Set ID		
	0	▼
	Proxy Address	
1	10.40.1.43:10060	UDP ▼
2		▼
3		▼
4		▼
5		▼
▼ Enable Proxy Keep Alive		
	Disable	▼

Figure 12: Configure the proxy IP address and transport type

7. Select **VoIP > SIP Definitions > General Parameters**.

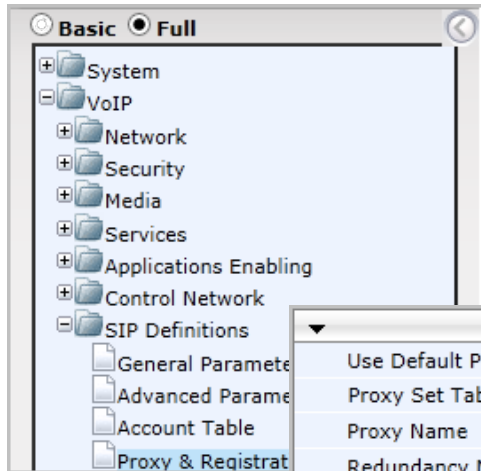


Parameter	Value
NAT IP Address	0.0.0.0
PRACK Mode	Supported
Channel Select Mode	By Dest Phone Number
Enable Early Media	Disable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	UDP
SIP UDP Local Port	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

- Make sure that *NAT IP Address* is **0.0.0.0**.
- Set *Enable Early Media* to **Disable**.
- Set *SIP Transport Type* to **UDP**.
- Set *SIP UDP Local Port* to **10060**.
- Set *SIP Destination Port* to **10060**.

Figure 13: Configure the general SIP parameters

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



- Set *Use Default Proxy* to **Yes**.
- Set *Enable Registration* to **Enable**.
- Set *Registrar IP Address* to **10.40.1.43**. (The MaxCS public IP address.)
- Set *Registrar Transport Type* to **UDP**.
- Set *Registration Time* to **100**.
- Set *Re-registration Timing [%]* to **50**.


Use Default Proxy	Yes
Proxy Set Table	
Proxy Name	
Redundancy Mode	Homing
Proxy IP List Refresh Time	60
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	Routing Table
SIP ReRouting Mode	Standard Mode
Enable Registration	Enable
Registrar Name	
Registrar IP Address	10.40.1.43
Registrar Transport Type	UDP
Registration Time	100
Re-registration Timing [%]	50

Figure 14: Configure the proxy and registration settings

9. Select **VoIP > Coders and profiles > Coders**. Make sure that both G.711U-law and G.729 are in the list.

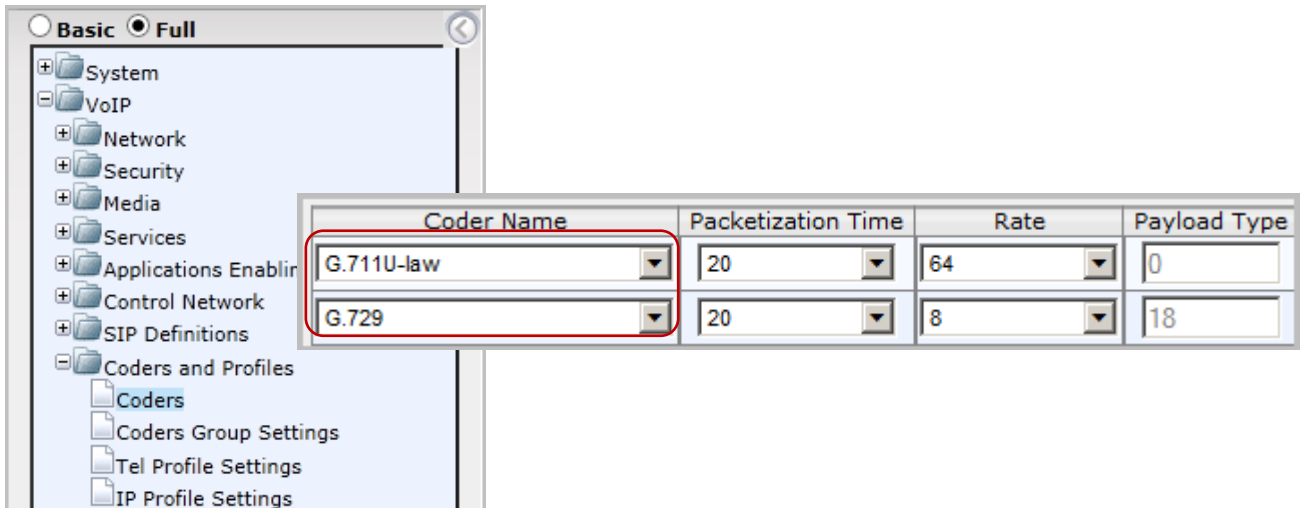
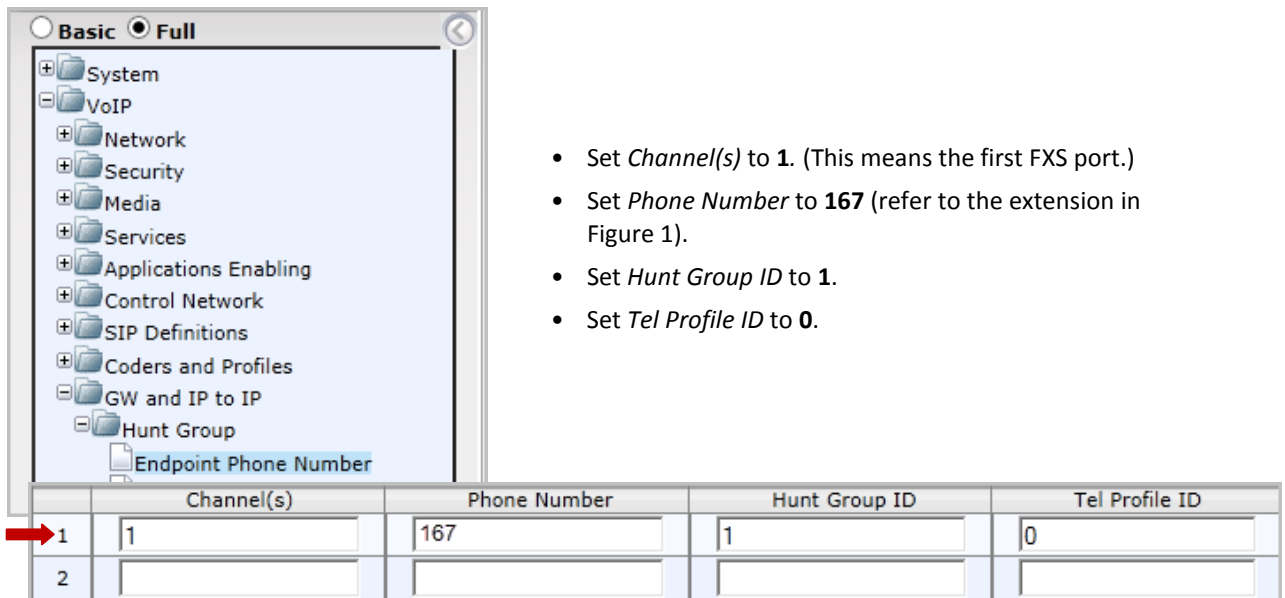


Figure 15: Confirm that G.711U-law and G.729 are listed in Coders and Profiles

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



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

Figure 16: Configure the endpoint phone extension



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

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

Hunt Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID
1	By Dest Phone Number	Per Endpoint	
2			

Figure 17: Configure hunt group settings

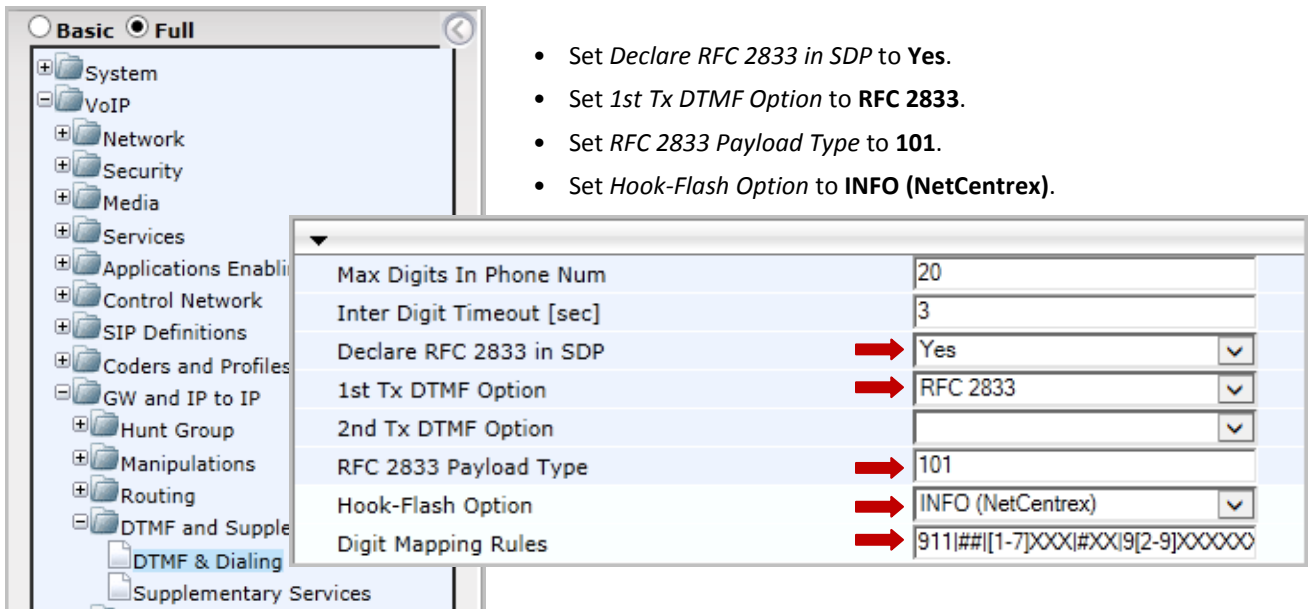
12. Select **VoIP > GW and IP to IP > Routing > IP to Hunt Group Routing**. Configure Entry 1 as follows:

- Set the first five fields, *Dest. Host Prefix*, *Source Host Prefix*, *Dest. Phone Prefix*, *Source Phone Prefix*, and *Source IP Address*, to *****.
- Set *Hunt Group ID* to **1**. (This assumes that the FXS port's hunt group ID is 1.)

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID
1	*	*	* x	*	*		1
2	*	*	*	*	*		

Figure 18: Configure hunt group routing rules

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



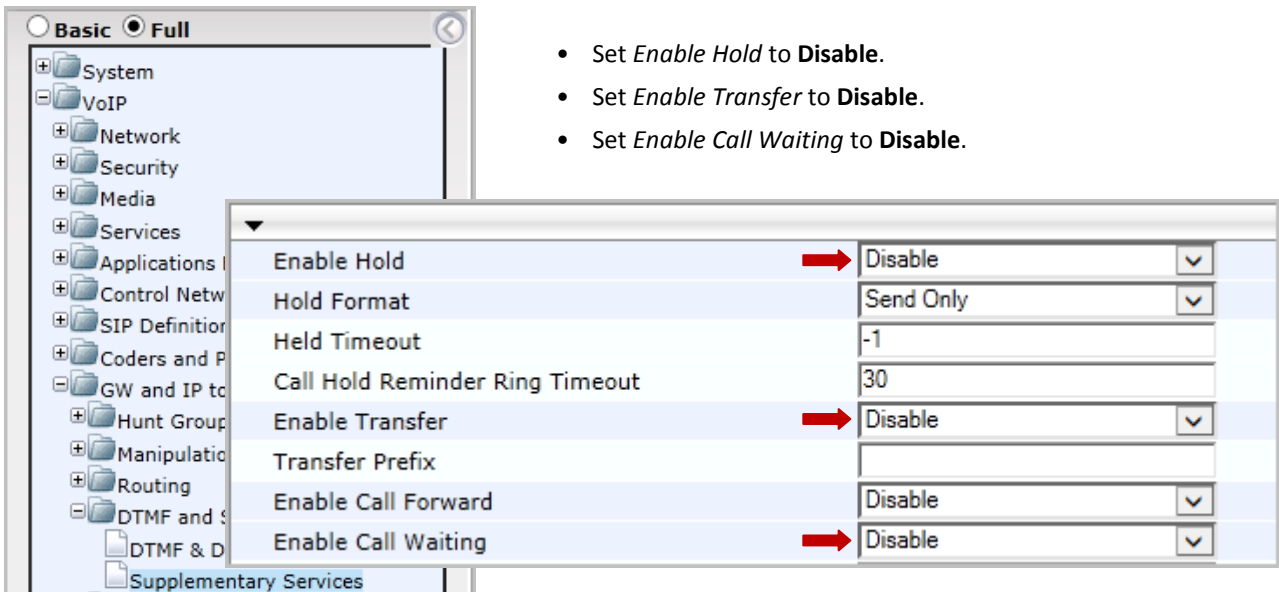
Max Digits In Phone Num	20
Inter Digit Timeout [sec]	3
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	
RFC 2833 Payload Type	101
Hook-Flash Option	INFO (NetCentrex)
Digit Mapping Rules	911 ## [1-7]XXX #XX 9[2-9]XXXXXX 91[2-9]XXXXXXXX XX.T

- Set *Declare RFC 2833 in SDP* to **Yes**.
- Set *1st Tx DTMF Option* to **RFC 2833**.
- Set *RFC 2833 Payload Type* to **101**.
- Set *Hook-Flash Option* to **INFO (NetCentrex)**.

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

Figure 19: Configure DTMF and dialing settings

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



Enable Hold	Disable
Hold Format	Send Only
Held Timeout	-1
Call Hold Reminder Ring Timeout	30
Enable Transfer	Disable
Transfer Prefix	
Enable Call Forward	Disable
Enable Call Waiting	Disable

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

Figure 20: Disable hold, transfer, and call waiting

15. 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.)
- Set *MWI Display* to **Enable**.
- Set *Subscribe to MWI* to **Yes**.
- Set *MWI Server IP Address* to **10.40.1.43** (The MaxCS Server IP address, as shown in Figure 1.)
- Set *MWI Transport Type* to **UDP**.

▼ Message Waiting Indication (MWI) Parameters	
Enable MWI	Enable
MWI Analog Lamp	Disable
MWI Display	Enable
Subscribe to MWI	Yes
MWI Server IP Address	10.40.1.43
MWI Server Transport Type	UDP

Figure 21: Configure message waiting indication settings

16. On the same page, configure the following setting in the *Conference* section:

- Set *Enable 3-Way Conference* to **Disable**.

▼ Conference	
⚡ Enable 3-Way Conference	Disable
Establish Conference Code	!
Conference ID	conf

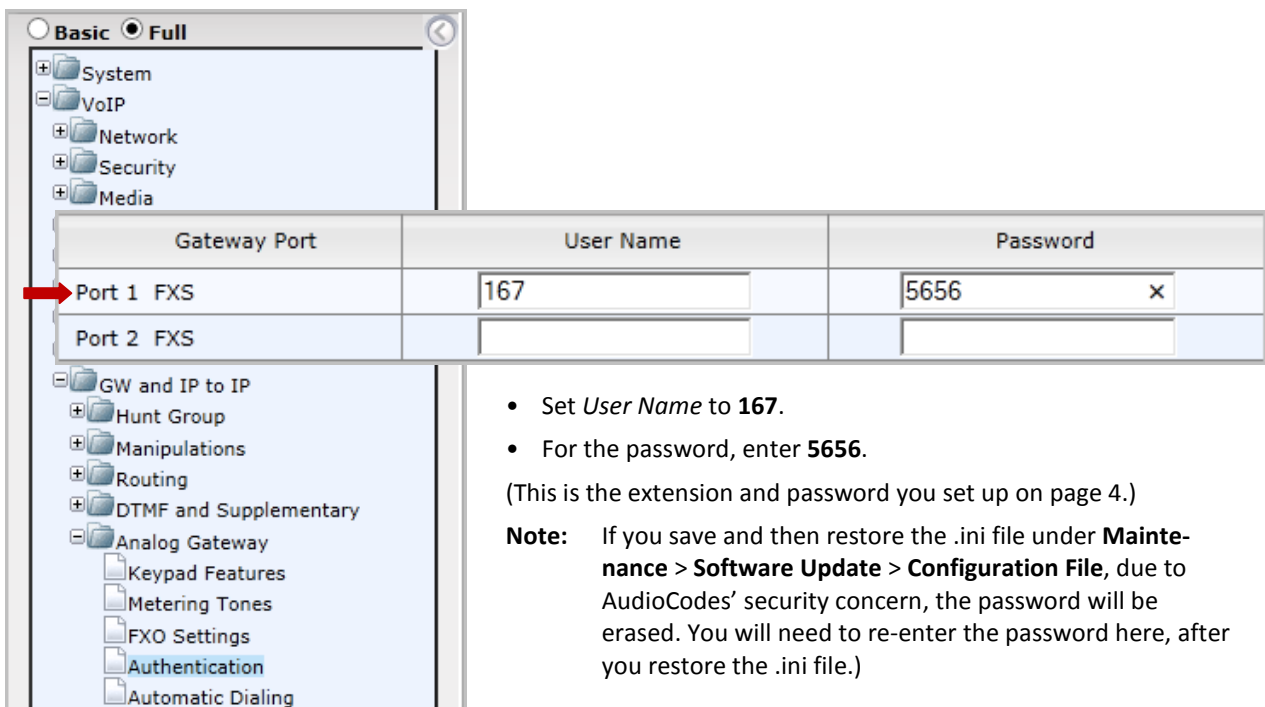
Figure 22: Disable 3-way conference setting

17. Select **VoIP > GW and IP to IP > Analog Gateway > Caller ID Permissions**. Enable for Caller ID for all ports.

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

Figure 23: Enable Caller ID

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



Gateway Port	User Name	Password
Port 1 FXS	167	5656 x
Port 2 FXS		

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

(This is the extension and password you set up on page 4.)

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

Figure 24: Configure authentication settings

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

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

(extension number)xatgnemx(E911 LID)

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

Figure 25: Enter E911 Location ID for each port

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 MaxAdministrator) in the *Password* field (refer to the chapter *Setting Up IP Extensions* in the *MaxCS 7.0 Update 1 Administration Manual*).

For details on configuring E911 Location IDs for IP phones, refer to the chapter *Location-Based E911 (for Relocation)* in the *MaxCS 7.0 Update 1 Administration Manual*.

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

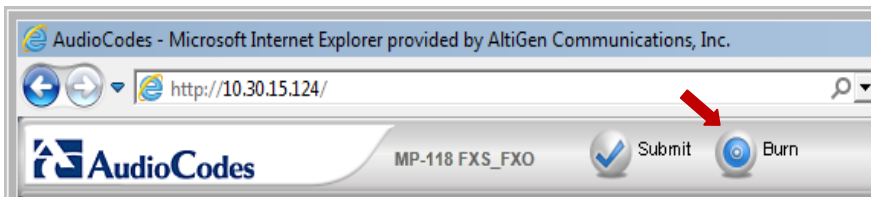


Figure 26: Click the **Burn** button.

- Reboot your AudioCodes device so that all of the changes take effect. Then proceed to the next section to verify your settings.

Verification

Now that you have configured both MaxCS 7.0 and AudioCodes, 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.

2. Click the **Register** button.

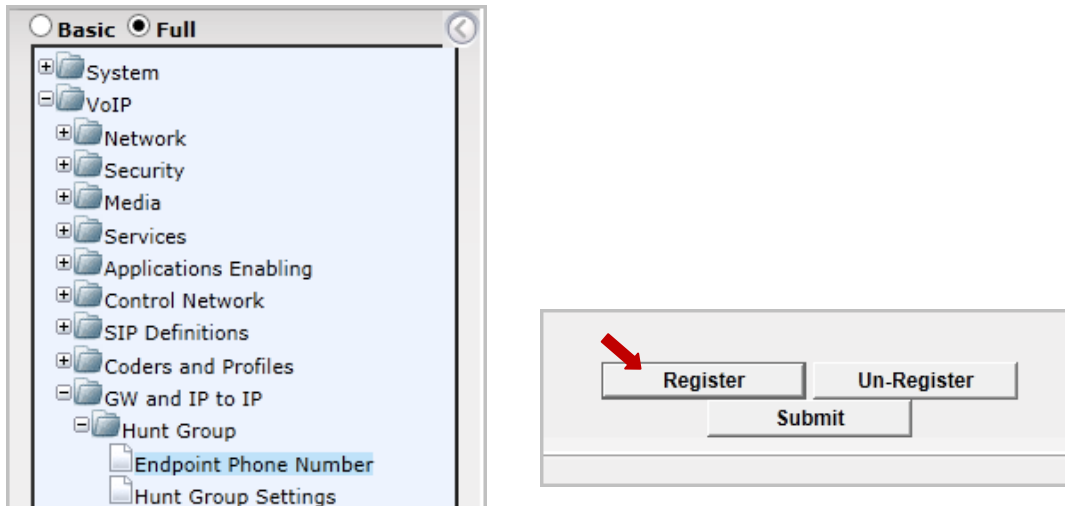


Figure 27: Register the phone extension

3. Log into MaxCS 7.0 MaxAdministrator. Open the Extension view; extension 167 should be listed there.

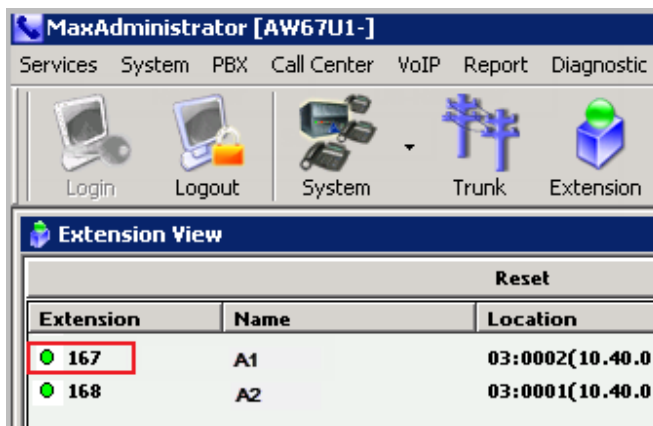


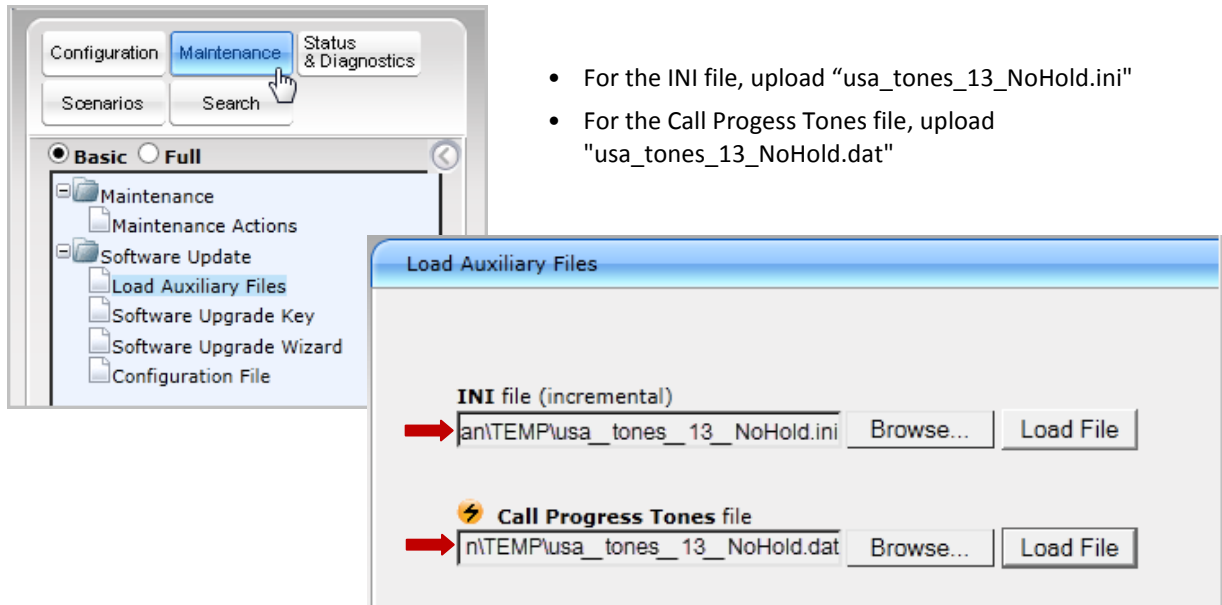
Figure 28: The extension appears in Extension View in MaxCS Administrator

4. Attach an analog phone to the first FXS port and make a few calls to test that everything works correctly.
5. If everything works correctly, return to the AudioCodes configuration tool and click the **Maintenance** button above the menu (see the next figure). Then select **Software Update > Load Auxiliary Files**.

At this point, you will upload the two files that came with this configuration guide, as mentioned in the section [Prerequisites](#) on page 4.

- usa_tones_13_NoHold.ini
- usa_tones_13_NoHold.dat

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



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

Figure 29: Upload the INI and DAT files

6. Submit this last change. On the toolbar, click **Burn**.
7. Restart the AudioCodes gateway.

The INI File

Here are the contents of the usa_tones_13_NoHold.ini file.

```
[NUMBER OF CALL PROGRESS TONES]
Number of Call Progress Tones=13

#Dial tone
[CALL PROGRESS TONE #0]
Tone Type=1
Tone Form =1
Low Freq [Hz]=350
High Freq [Hz]=440
Low Freq Level [-dBm]=13
High Freq Level [-dBm]=13
First Signal On Time [10msec]=200
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0

#Dial tone
[CALL PROGRESS TONE #1]
Tone Type=1
Tone Form =1
```

```
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=10
High Freq Level [-dBm]=32
First Signal On Time [10msec]=300
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0

#Ringback
[CALL PROGRESS TONE #2]
Tone Type=2
Tone Form =2
Low Freq [Hz]=440
High Freq [Hz]=480
Low Freq Level [-dBm]=19
High Freq Level [-dBm]=19
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=200
Second Signal Off Time [10msec]=400

#Ringback
[CALL PROGRESS TONE #3]
Tone Type=2
Tone Form =2
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=16
High Freq Level [-dBm]=32
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=100
Second Signal Off Time [10msec]=300

#Busy
[CALL PROGRESS TONE #4]
Tone Type=3
Tone Form =2
Low Freq [Hz]=480
High Freq [Hz]=620
Low Freq Level [-dBm]=24
High Freq Level [-dBm]=24
First Signal On Time [10msec]=50
First Signal Off Time [10msec]=50
Second Signal On Time [10msec]=50
Second Signal Off Time [10msec]=50

#Busy
```

[CALL PROGRESS TONE #5]
Tone Type=3
Tone Form =2
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=20
High Freq Level [-dBm]=32
First Signal On Time [10msec]=50
First Signal Off Time [10msec]=50
Second Signal On Time [10msec]=50
Second Signal Off Time [10msec]=50

#Reorder tone
[CALL PROGRESS TONE #6]
Tone Type=7
Tone Form =2
Low Freq [Hz]=480
High Freq [Hz]=620
Low Freq Level [-dBm]= 24
High Freq Level [-dBm]=24
First Signal On Time [10msec]=25
First Signal Off Time [10msec]=25
Second Signal On Time [10msec]=25
Second Signal Off Time [10msec]=25

#Reorder tone
[CALL PROGRESS TONE #7]
Tone Type=7
Tone Form =2
Low Freq [Hz]=1400
High Freq [Hz]=0
Low Freq Level [-dBm]= 24
High Freq Level [-dBm]=32
First Signal On Time [10msec]=10
First Signal Off Time [10msec]=10
Second Signal On Time [10msec]=10
Second Signal Off Time [10msec]=10
Third Signal On Time [10msec]=10
Third Signal Off Time [10msec]=10

#Confirmation tone
[CALL PROGRESS TONE #8]
Tone Type=8
Tone Form =2
Low Freq [Hz]=350
High Freq [Hz]=440
Low Freq Level [-dBm]=20
High Freq Level [-dBm]=20
First Signal On Time [10msec]=10



First Signal Off Time [10msec]=10
Second Signal On Time [10msec]=10
Second Signal Off Time [10msec]=10

#Call Waiting Tone

[CALL PROGRESS TONE #9]
Tone Type=9
Tone Form =2
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=20
High Freq Level [-dBm]=32
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=30
Second Signal Off Time [10msec]=500

#Stutter Dial tone

[CALL PROGRESS TONE #10]
Tone Type=15
Tone Form =1
Low Freq [Hz]=350
High Freq [Hz]=480
Low Freq Level [-dBm]=13
High Freq Level [-dBm]=13
First Signal On Time [10msec]=500
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0

#Off Hook Warning Tone

[CALL PROGRESS TONE #11]
Tone Type=16
Tone Form =2
Low Freq [Hz]=1400
High Freq [Hz]=2600
Low Freq Level [-dBm]=0
High Freq Level [-dBm]=0
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=10
Second Signal Off Time [10msec]=10

#Call Waiting Ringback Tone

[CALL PROGRESS TONE #12]
Tone Type=17
Tone Form =2
Low Freq [Hz]=440
High Freq [Hz]=480



```
Low Freq Level [-dBm]=19
High Freq Level [-dBm]=19
First Signal On Time [10msec]=120
First Signal Off Time [10msec]=40
Second Signal On Time [10msec]=40
Second Signal Off Time [10msec]=400

;#Hold Tone
;[CALL PROGRESS TONE #13]
;Tone Type=23
;Tone Form =2
;Low Freq [Hz]=1400
;High Freq [Hz]=0
;Low Freq Level [-dBm]=15
;High Freq Level [-dBm]=32
;First Signal On Time [10msec]=0
;First Signal Off Time [10msec]=0
;Second Signal On Time [10msec]=40
;Second Signal Off Time [10msec]=500

# Ringing signal
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=3

[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=30
First Ring On Time [10msec]=200
First Ring Off Time [10msec]=400

[Ringing Pattern #1]
Ring Type=1
Freq [Hz]=20
First Ring On Time [10msec]=100
First Ring Off Time [10msec]=50
Second Ring On Time [10msec]=100
Second Ring Off Time [10msec]=300

[Ringing Pattern #2]
Ring Type=2
Freq [Hz]=20
First Burst Ring On Time [10msec]= 50
First Burst Ring Off Time [10msec]= 50
Second Burst Ring On Time [10msec]= 50
Second Burst Ring Off Time [10msec]= 50
Third Ring On Time [10msec]=100
Third Ring Off Time [10msec]=300
```



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 device 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

AltiGen does not provide general configuration support for AudioCodes products. For general configuration information, 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