

MAX Communication Server Release 7.5

> TelePacific SIP Trunk Configuration Guide

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Intended audience: AltiGen Authorized Partners



# **About This Guide**

This guide is provided for AltiGen Partners who are provisioning TelePacific SIP Trunks for customers for MaxCS 7.5.

### Requirements

• You must be running MaxCS Release 7.5.

# **Create a New Codec Profile**

Configure a Codec Profile for the TelePacific SIP trunks in Enterprise Manager.

- 1. In MAXCS Administrator, select **VoIP** > **Enterprise Network Manager**.
- 2. Click the **Codec** button on the Quick Launch bar (this is different from the IP Codec tab).
- 3. Create a new profile; click **Add** and create a profile with the following parameters:
  - For the name, enter Telepacific SIP Trunks
  - Set the *Selected Codec* to use **G.711 Mu-Law** first and **G.729** second (use the **Add**, **Remove**, **Up**, and **Down** buttons as needed)
  - Set DTMF Delivery to RFC 2833
  - Set SIP Early Media to Enable
  - Set SIP Transport to UDP

Name:	Telepacific SIP Trunks			
	Selected Codec		Available Codec	
	G.729 G.729	< Add	G.711 A-Law	
		Remove>	G.722	
Codec:				
		00		
		Down		
			Advanced	
DTMF Delivery	RFC 2833			
SIP Early Media	Enable			
SIP Transport	UDP			

4. Click **Advanced** and make sure that your settings match those in the next figure.



vanced Setting				
tter Buffer Range				
G.711 Jitter Buffer Range:	Min	10 <b>.</b> ms	Max	100 <b>)</b> ms
G.722 Jitter Buffer Range:	Min	10 🏶 ms	Max	100 <b>)</b> ms
G.723 Jitter Buffer Range:	Min	30 🌒 ms	Мах	480 🌲 ms
G.729 Jitter Buffer Range:	Min	10 🏶 ms	Max	480 🌲 ms
TP Package Length				
G.711 RTP Packet Leng	th (ms)	20		•
G.722 RTP Packet Leng	th (ms)			
G 729 RTP Packet Leng	th (ms)	20		-

# **Create a new SIP Trunk Profile**

Next, create a new SIP Trunk Profile.

- 1. In MaxCS Administrator select **Boards**. Double-click **SIPSP**, select **SIP Trunk > SIP Trunk Configuration** > **SIP Trunk Profile**.
- 2. Click Add. Name the new profile Telepacific SIP Trunks.

rofiles	SIP Calling Number			
Default Skype SIP Trunks	SIP Protocol Field FROM Header			
Telepacific SIP Trunks	Carrier can accept any number			
	C Carrier can only accept Calling Number with minimum gig dig			
	C Carrier can only accept assigned numbers as Calling Number			
	Calling Number can be accepted by Carrier			
	From To Add			
	Edit			
	Use this as Calling Number if the Carrier cannot accept configured numbers			
	Use this as Calling Number if the Carrier cannot accept configured numbers			
	Use this as Calling Number if the Carrier cannot accept configured numbers			
	Use this as Calling Number if the Carrier cannot accept configured numbers			
	Use this as Calling Number if the Carrier cannot accept configured numbers			
	Use this as Calling Number if the Carrier cannot accept configured numbers Send Caller Name Enable Standard Record-Route Header Incoming DID Number field © To Header C Request URI Enable Fax Trunk Routing			
	Use this as Calling Number if the Carrier cannot accept configured numbers Send Caller Name Enable Standard Record-Route Header Incoming DID Number field To Header Request URI Enable Fax Trunk Routing Fax User Name			



- 3. Modify the new SIP Trunk profile, specifying the following parameters, and click **OK**.
  - Set SIP Protocol Field to FROM Header
  - Select Carrier can accept any number
  - Select Send Caller Name
  - Set Incoming DID Number to **To Header**
  - Make sure that the Enable Fax Trunk Routing checkbox is not checked

#### **Configure Channels**

Configure channels to use the TelePacific SIP Trunk.

- 1. Select **Boards**. Double-click **SIPSP**, select **SIP Trunk > SIP Trunk Configuration**.
- 2. Select the first available SIP Trunk channel and click **Edit**.
- 3. Modify the parameters as follows, and click **Ok**.
  - For the SIP Server IP address, enter the IP address provided by TelePacific; this IP address is their SIP Trunk router
  - o For the User Name, enter the 10-digit billing telephone number of the SIP Trunk service
  - Leave the Password and Domain fields blank
  - o Set SIP Registration Period to **o** seconds
  - For the SIP Trunk Profile, select Telepacific SIP Trunks
  - Set the SIP Source Port and SIP Destination Port to 5060
  - Check the Enable Channel checkbox

SIP Trunk - Id=5, Log	ical Channel Id=77	×
SIP Server IP Address	192.168.20.239	
User Name	9492083799	
Password		
Domain		
SIP Register Period	0 Sec.	
SIP Trunk Profile	Telepacific SIP 💌	
SIP Source Port	5060 💌	
SIP Destination Port	5060	
🔽 Enable Channel		
	OK Cancel	

4. Select the first available SIP Trunk channel and click **Copy To...** 



- 5. Select all SIP channels that will be used for TelePacific SIP Trunks, and click **Ok**.
- 6. Click **Trunk Group Configuration**. Select the SIP server with the IP address that you entered in step 3 in this section.
  - For SIP Server Name, enter Telepacific SIP Trunks
  - Clear the Enable SIP OPTIONS check box
- 7. Click Ok.

		Str Hunk List	STD ODTIONS Chart
IP Server	Name	ID Channel No	SIP OPTIONS Client
92.168.20.239	Telepacific SIP	0 72 1 73 2 74 3 75 4 76 5 77 6 78 7 79 8 80 9 81 10 82 11 83	SIP Server Name Telepacific SIP Trunks Finable SIP OPTIONS SIP OPTIONS Interval 30 seconds Number of Retries 5 times Retry Interval 2 seconds

#### **Configure Inbound Routing**

Carriers send 10 digits as DNIS; configure your inbound routing rules accordingly.

#### **Test the SIP Trunks**

We recommend that you perform basic tests to confirm that the SIP Trunks are correctly configured; test inbound calls, outbound calls, and extension-to-extension calls.