TECHNICAL NOTE

> MAX 109

Attaching the MAX IP SIP Phone to a Cisco CallManager Switch

Description

As of the writing of this document the majority of Cisco IP Telephony phone systems do not support native SIP endpoints. As such, a SIP proxy becomes a required component before SIP endpoints will work in a Cisco IP Telephony environment. This document describes how to use the Asterisk@home open source software PBX as a SIP proxy server to allow the MAX IP phone to function in a Cisco IP Telephony environment.

Intended Audience

This document is intended for VOIP network administrators who have a basic understanding of SIP, IP Telephony, and Cisco CallManager. Additional skills in Linux is helpful.

Environment

The environment used when setting up this technical note was as follows:

- CallManager v4.0(2)a
- Asterisk@home v1.3
- Two Max IP Conference Phones

Differences in steps or supported functionality might exist depending on the version of CallManager/Asterisk@home you are using. Please see the **Helpful Web Sites & Resources** section of this document for additional resources.

Helpful Web Sites & Resources

- http://www.cisco.com Cisco Systems Web Site
- http://asteriskathome.sourceforge.net Asterisk@home home page.
- http://asteriskathome.sourceforge.net/handbook/ Quick Guide / Documentation for Asterisk@home
- http://www.voip-info.org/wiki/view/Asterisk+Cisco+CallManager+Express+Integration integrate Asterisk with CallManager Express
- http://www.voip-info.org/wiki-Asterisk+Cisco+CallManager+Integration integrate Asterisk with Cisco CallManager 3.x using H.323 (SIP not available) or with 4.0 using a SIP trunk
- http://www.xten.com SIP Softphone provider



Quick Steps

- 1. Set up Asterisk@home server
- 2. Set up MAX IP SIP Endpoints on Asterisk@home
- 3. Test dial between SIP endpoints (validate Asterisk@home setup)
- 4. Set up CallManager SIP Trunk
- 5. Set up CallManager Route Patterns and point the pattern to the SIP Trunk
- 6. Set up Asterisk@home SIP Trunk
- 7. Set up Asterisk@home dial plan and point the route pattern to the SIP Trunk
- 8. Final Test

Step 1: Set up Asterisk@home proxy system

Asterisk@home is a free and easy to install Linux based software PBX that can be quickly configured as a SIP proxy.

- 1.1 Download the latest ISO from http://asteriskathome.sourceforge.net and burn it to a CD.
- 1.2 After the CD has been created, insert the CD into the system that will become the SIP proxy and reboot the machine. When prompted, press **Enter** to start the installation.

WARNING – This will erase all data on the local hard drive!!!

- 1.3 A base Linux system (CentOS Distribution) is auto installed and the CD is ejected. Please be patient while the system finishes its installation and reboots for the first time.
- 1,2 Log into the Asterisk@home system. The default username and password are:

(Username: root, Password: password)

- 1.3 (RECOMMENDED) Change your password by typing passwd. Failure to do so will place your system at risk for security breaches.
- 1.4 Set up an IP address for your SIP Proxy.

Type netconfig at the command line. This allows you to set a static IP address for your Asterisk@home box. (Linux experience is helpful if you desire to set up advanced networking configurations.)

If you prefer to use DHCP, please ensure that your lease is permanent and take note of the address assigned during the local system login.

- 1.5 After you have configured your network card, reboot the system by typing reboot.
- 1.6 Once restarted, verify that you can get to the system from a web browser on your network. If not, you will need to troubleshoot the SIP server network settings.



Figure 1

Step 2: Set up SIP endpoints

2.1 Connect to your Asterisk@home box using a web browser and then click on the Asterisk Management Portal (see Figure 1). You need to login using the default username and password listed below:

(user: maint, pass: password)

sterisk Management ortal	🔄 Asterisk Managem	ent Portal - Microsoft Internet	Explorer provided by Cl	earOne Communications In	ю.	_ @ <u>×</u>
	File Edit View Fav	orites Tools Help				A.
	🕒 Back • 🕥 •	💌 😰 🏠 🔎 Search 🥎	💦 Favorites Media 📢	8 🙆 - 👗 🕞 🗾		
	Address a http://10.0.0	0.1/admin/config.php?display=extensi	ons&tech=sip			Go Links »
		ANP	Asterisk Management Portal	Maintenance	Setup • Reports • Panel	4
		Incoming Calls	Add SIP Extens	sion	Add Extension	
		Extensions	Add Extension		<101>	
		Ring Groups			<201>	
		Queues	Extension Number:		1	
		Digital Receptionist	Display: Name:			=
		Trunks	bibpidy Name.			
		Inbound Routing	Extension Options			
		Outbound Routing				
		On Hold Music	Outbound CID:			
		System Recordings	Record Incoming:	On Demand 🚩		
		Backup & Restore	Record Outgoing:	On Demand ≚		
		General Settings	Device Options			
			secret			
			dtmfmode	rfc2833	j	
			Voicemail & Directo	ory: Disabled 💙		
	A 1					Internet
	te start	Asterisk Management	rovable Disk (E:)	cument 1 - Microsof		Tincerner

- 2.2 Click Setup -> Extensions -> SIP Extension.
- 2.3 Assign an extension number for your endpoint and type in a password in the secret field for registration. Scroll to the bottom of the screen and click submit.
- 2.4 (OPTIONAL) Enter a display name for the extension.
- 2.5 Configure your DHCP server to have Option 66 enabled to direct the MAX IP to a tftp server where the MAX IP phone configuration and dial plan files have been stored for download by the phone on boot up.

2.6 Set up the MAX IP SIP endpoint for this extension.

Using the MAX IP configuration file in Appendix 1 of this document, edit the following lines of the file for your specific needs:

Line 5 <localnum> xxxx </localnum> – This is the extension number of the MAX IP phone.

Line 26 <sip_username > name </sip_username > - Username set up in the Asterisk server.

Line 27 <sip_password> password </sip_password> - Password set up in the Asterisk server.

Line 29 <sip_proxy_server> 0.0.0.0 </sip_proxy_server> - IP address of the Asterisk server.

Save the file with the name CIMAXIP_macaddress.txt. The *mac address* will be on the base of the MAX IP phone. Place this file on your TFTP server

2.7 Using the default dial plan in Appendix 2, edit the digitmap match line so that the DIAL_STRING line contains the IP address of the Asterisk server. Save this file as clmaxlavdial.txt and place it on your TFTP server. This dial plan will be used by all the MAX IP phones on the network.

Repeat steps 2.2 thru 2.4 and 2.6 for each additional phone.

Step 3: Test between SIP endpoints

NOTE: Testing requires at least two endpoints. If you only have one Max IP phone you will want to get a SIP softphone set up for testing. X-Ten offers a free softphone that can be used for this purpose. Download the softphone at http://www.xten.com and then set it up for use with Asterisk@home.

- 3.1 Dial the extension and verify that ring tones are generated on near and far sides.
- 3.2 Answer the phone and test to ensure you have bi-directional audio. (One way audio signifies an error on the network and/or systems configuration.)

If you have problems during this test, do not move on until the Asterisk@home SIP system is able to set up and tear down SIP endpoint-to-endpoint calls correctly.

Step 4: Set up CallManager SIP trunk

- 4.1 In Cisco CallManager administration, go to Device -> Trunk.
- 4.2 Add a Trunk.
- 4.3 Set up as shown in Figure 2 and described below.

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> Figure 2

Cisco CallManager Trunk Configuration

Cisco CallManager A	dministration		Cisco Systems
Trunk Configura	tion		Add a New Trunk Back to Find/List Trunk Dependency Records
Product: SIP Trunk Device Protocol: SIP Status: Ready Update Delete Reset Tru	nk		
Device Information			
Device Name*	Asterisk		
Description	Asterisk		
Device Pool*	SLC-DPA	•	
Media Resource Group List	MRGL-B	-	
Location	SLC	•	
AAR Group	< None >	-	
🗹 Media Termination Point Red	juired		
Destination Address*	10.101.11.232		
Destination Address is an SR	v		
Destination Port	5060		
Incoming Port*	5060		
Outgoing Transport Type*	UDP	-	
Preferred Originating Codec*	711ulaw	-	
Call Routing Information			
Inbound Calls			
Significant Digits*	All	-	
Connected Line ID Presentation*	Default	3	
Connected Name Presentation*	Default	•	
Calling Search Space	SLC_Unrestricted	•	
AAR Calling Search Space	SLC_Unrestricted	-	

Device Name: As desired.

Description: As desired.

Device Pool, Media Resource Group List, Location, AAR Group: Depends on CallManager Environment.

Media Termination Point Required: Checked.

Destination Address: IP Address of the Asterisk@home proxy server.

Destination Address is an SRV: Not checked.

Destination Port: Defaults to 5060 but may be changed depending on environment. Make sure port matches on Asterisk@home configuration.

Incoming Port: Defaults to 5060 but may be changed depending on environment. Make sure port matches on Asterisk@home configuration.

Outgoing Transport Type: UDP

Preferred Originating Codec: Defaults to g711ulaw. Settings must match between Asterisk@home and CallManager.

Call Routing Information: Depends on CallManager Environment. **Outbound Calls:** Depends on CallManager Environment.



Step 5: Set up CallManager route pattern and point the pattern to the SIP trunk

- 5.1 In CallManager Administration, go to RoutePlan -> RoutePattern/HuntGroup.
- 5.2 Add a Route Pattern.
- 5.3 Set up the route pattern as show in Figure 3 and described below.

Cisco CallManager A For Cisco IP Telephony Solutions	Administration
Route Pattern/H Configuration	Add a New Route Pattern/Hunt Pilot Back to Find/List Route Patterns and Hunt Pilots
Route Pattern/Hunt Pilot: 35	2X
Status: Ready Note: Any update to this Route Patter	rn or Hunt Pilot automatically resets the associated gateway or Route/Hunt List
Copy Update Delete	
Pattern Definition	
Route Pattern/Hunt Pilot*	352×
Partition	SLC_Phones
Description	
Numbering Plan*	North American Numbering Plan
Route Filter	< None >
MLPP Precedence	Default
Gateway or Route/Hunt List*	Asterisk (Edit)
Route Option	Route this pattern
	O Block this pattern - Not Selected -
Provide Outside Dial Tone	Allow Overlap Sending Urgent Priority

Route Pattern/Hunt Pilot: Enter the route pattern that should be forwarded to the Asterisk Proxy for service.

Partition: Depends on CallManager Environment.
Description: Optional.
Numbering Plan: Depends on CallManager Environment.
Route Filter: Depends on CallManager Environment.
MLPP Precedence: Depends on CallManager Environment.
Gateway or Route/Hunt List: Choose the Asterisk Trunk.
Route Option: Route this Pattern.
Other Settings: Depends on CallManager Environment.

Step 6: Set up Asterisk@home SIP trunk

- 6.1 Login to AMP at http://hostname/admin/.
- 6.2 Click on Setup -> Trunks.
- 6.3 Add SIP Trunk.
- 6.4 Configure with the settings shown in Figure 4 and described below.

> Figure 3

Cisco CallManager Route Pattern/Hunt Pilot Configuration

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> Figure 4

SIP Trunk Configuration Settings

Outgoing Settin	gs	
Trunk Name:	caliman01	
PEER Details:		
allow=ulaw&alaw canreinvite=yes context=default disallow=all host=172.16.0.1 nat=no		×
qualify=yes type=friend		
Incoming Settin	gs default	
USER Details:		
allow=ulaw£alaw canreinvite=yes context=default disallow=all host=172.16.0.1 nat=no qualify=yes tume=f=iend		
cype=rriena		

The host IP address in the configuration refers to the IP address of the CallManager receiving (peer) or sending (incoming). The other settings may vary based on individual CallManager/Asterisk configurations.

6.5 Repeat 6.3 & 6.4 for each CallManager in the cluster.

Registration

Step 7: Set up Asterisk@home dial plan

- 7.1 Click on Outbound Routing -> Add Route.
- 7.2 Add Route.
- 7.3 Configure Dial Patterns to meet dial plan requirements. (The example shown in Figure 5 sends all digits after 9 to the SIP Trunk.)

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> Figure 5

Configuring Dial Patterns

Incoming Calls			Add Route
Extensions	Ealt Rou	te	0 Outside9 🤹
Ring Groups	Delete Route Out	tside9	1 Default 🍠
Queues			
Digital Receptionist	Route Name:	Outside9 Rename	
Trunks	Route Password		
Outbound Routing	Dial Patterns		
DID Routes			
On Hold Music		9. 💆	
System Recordings			
Backup & Restore		1	
General Settings		Clean & Remove duplicates	1
	Insert:	Pick pre-defined patterns	
	Trunk Sequence		
	0	SIP/callman01 💌 🗙	
		SIP/callman02 • Add	
		Submit Changes	-

Step 8: Test, test, test

8.1 Test until satisfied.

Appendix 1: Sample SIP Configuration File

<C1MAXSIPCONFIG> <username> clearone </username> <password> maxiav </password> <ringtone> 1 </ringtone> <localnum> 000 </localnum> <helpline num> 0 </helpline num> <allow reboot in call> 0 </allow reboot in call> <mute ringtone> 0 </mute ringtone> <dialplan> C1MAX1AVDIAL.txt </dialplan> <timezone> 5 </timezone> <SNTP_server_1> 0.0.0.0 </SNTP_server_1> <SNTP server 2> 0.0.0.0 </SNTP server 2> <DSCP TOS BITS> 0 </DSCP TOS BITS> <speed_dial_index 0> 0 </speed_dial_index 0> <speed dial index 1> 0 </speed dial index 1> <speed dial index 2> 0 </speed dial index 2> <speed dial index 3> 0 </speed dial index 3> <speed dial index 4> 0 </speed dial index 4> <speed dial index 5> 0 </speed dial index 5> <speed dial index 6> 0 </speed dial index 6> <speed dial index 7> 0 </speed dial index 7> <speed dial index 8> 0 </speed dial index 8> <speed dial index 9> 0 </speed dial index 9> <adjust_dst> 1 </adjust_dst> <use sipauth> 1 </use sipauth> <sip username> name </sip username> <sip_password> password </sip_password> <sip proxy enable> 1 </sip proxy enable> <sip proxy server> 0.0.0.0 </sip proxy server> <sip proxy port> 5060 </sip proxy port> <outbound sip proxy enable> 1 </outbound sip proxy enable> <outbound sip proxy> 0.0.0.0 </outbound sip proxy> <outbound_proxy_port> 5060 </outbound_proxy_port> <sip register timetout> 3600 </sip register timetout> <sip transport> 0 </sip transport> <sip_udp_port> 5060 </sip_udp_port> <sip tcp port> 5060 </sip tcp port> <dtmf_relay_enable> 1 </dtmf_relay_enable> <dtmf relay payload> 96 </dtmf relay payload> <g711ulaw priority> 255 </g711ulaw priority> <g711Alaw_priority> 254 </g711Alaw_priority> <g729ab priority> 250 </g729ab priority> <g7231 53 priority> 240 </g7231 53 priority> <g7231 63 priority> 245 </g7231 63 priority> </C1MAXSIPCONFIG>



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Appendix 2: Sample Dial Plan

<C1DIALPLAN> <SYSCONFIG DIALTIME="120000" FIRST_DIGIT_WAIT="30000" INTER DIGIT WAIT="15000" TERMINATION DIGIT="#"/> <DIGITMAP MATCH="+&" MIN DIGITS="1" MAX DIGITS="44" STRIP_FIRST_DIGITS="0" ADD_PREFIX_AFTER_STRIP="" DIAL_STRING="+&@0.0.0.0"/> </C1DIALPLAN>

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