ClearOne

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.

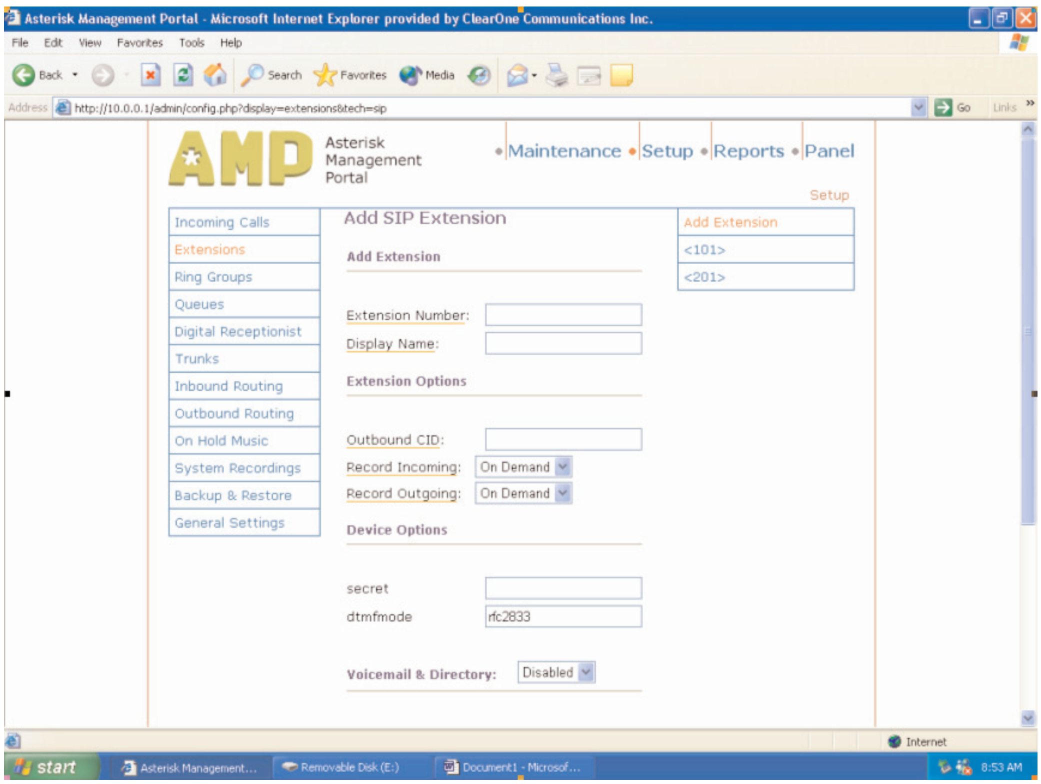
Step 2: Set up SIP endpoints

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 user-name and password listed below:

(user: maint, pass: password)

> **Figure 1**

Asterisk Management Portal



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 `C1MAXIP_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 `c1max1avdial.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

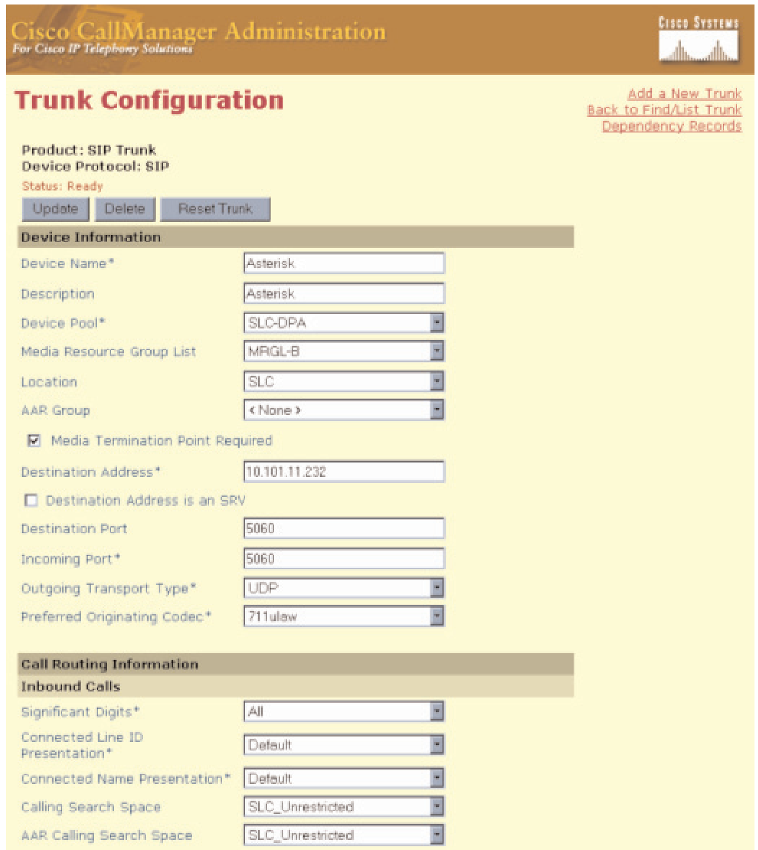
4.1 In Cisco CallManager administration, go to Device -> Trunk.

4.2 Add a Trunk.

4.3 Set up as shown in Figure 2 and described below.

> Figure 2

Cisco CallManager Trunk Configuration



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

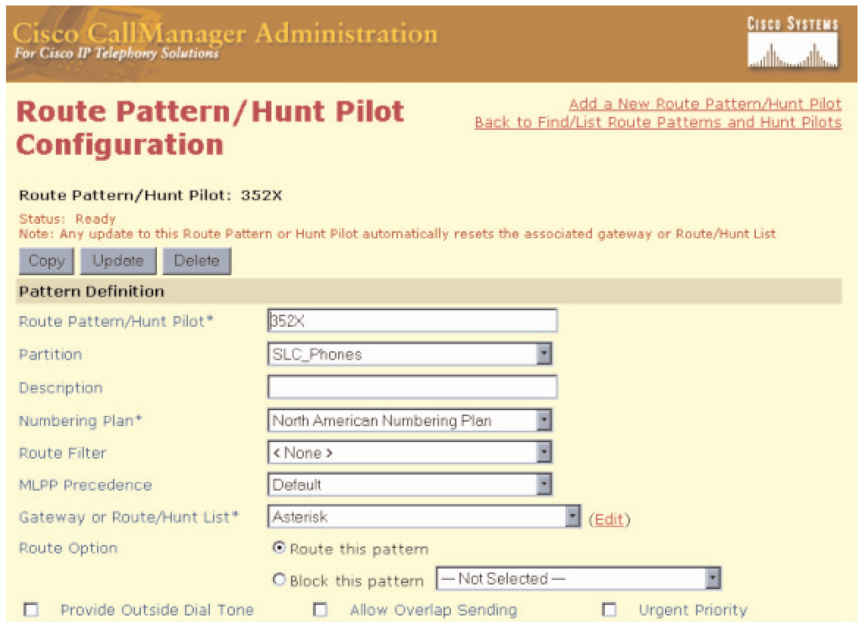
5.1 In CallManager Administration, go to RoutePlan -> RoutePattern/HuntGroup.

5.2 Add a Route Pattern.

5.3 Set up the route pattern as show in Figure 3 and described below.

> **Figure 3**

Cisco CallManager Route Pattern/Hunt Pilot Configuration



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/.

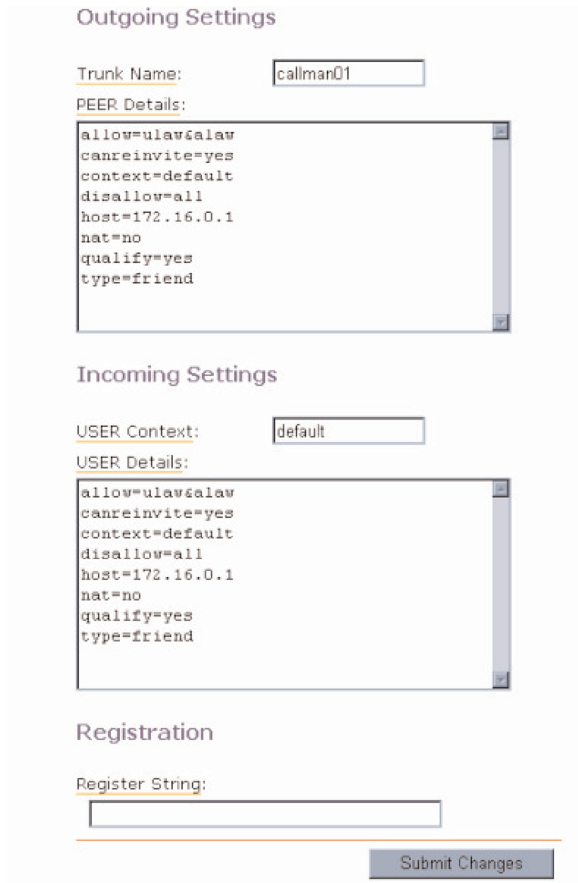
6.2 Click on Setup -> Trunks.

6.3 Add SIP Trunk.

6.4 Configure with the settings shown in Figure 4 and described below.

> **Figure 4**

SIP Trunk Configuration Settings



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.

Step 7: Set up Asterisk@home dial plan

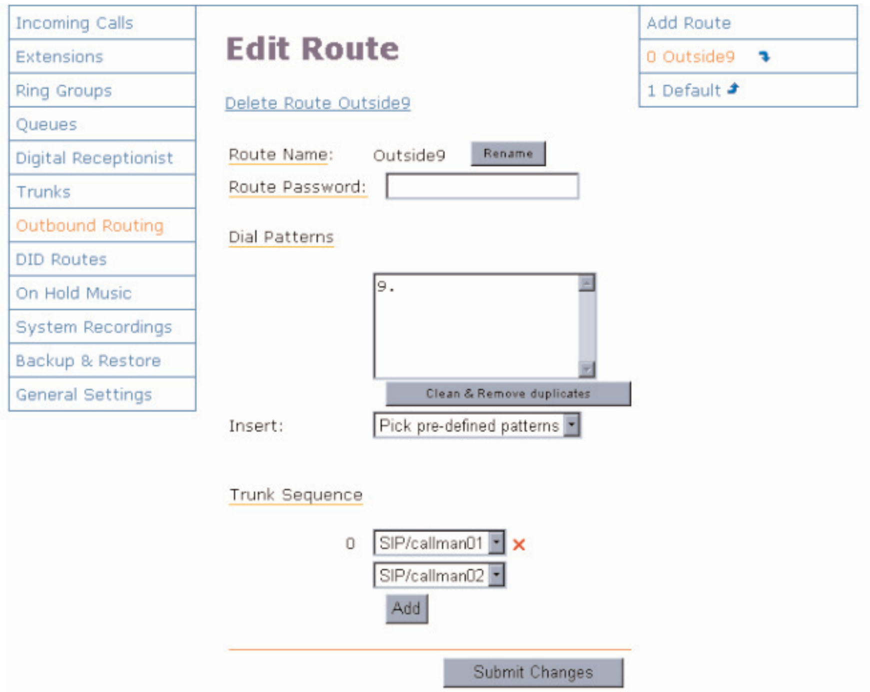
7.1 Click on Outbound Routing -> Add Route.

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.)

> **Figure 5**

Configuring Dial Patterns



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>
```

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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