

Configuring the SPA112 ATA to operate with the CUCM

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Scenario:

You have a Cisco Unified Communications Manager (CUCM) system and want to configure a SPA112 analog telephone adaptor (ATA) to register to the CUCM so that you can use up to two analog phones or similar FXS devices with the CUCM.

Overview:

In this application note, I have a Cisco Business Edition 6000 (informally also known as the BE6000 or BE6k). My BE6000 is running CUCM 9.0.1.10000-37.

I have a SPA112 ATA running 1.3.1(003) firmware and an analog phone connected to the PHONE 1 port of the SPA112.

1. The CUCM has a 4-digit internal dial plan starting with 1.
2. The CUCM is already configured and has other users and phones configured and able to make calls with each other.
3. The SPA112 ATA is starting from a factory defaulted state.

Configuring the CUCM

Configuring the CUCM is a discipline unto itself. This is why folk work so hard at acquiring their CCIE Voice certification. This application note does not address important topics such as security and optimal configuration which are mandatory for live deployment.

These instructions must be considered to be for internal and secure lab-use proof-of-concept only.

Overview

The following CUCM administration tasks to prepare the CUCM for connecting a SPA112 are described in detail:

1. Define a CUCM user that will be associated with the SPA112 Line1
2. Define a third-party SIP phone for the SPA112 Line1
3. Associate a Line / DN (directory number) to the previously defined SPA112 Line1 third-party SIP device
4. Associate the SPA112 Line1 user with the SPA112 Line1 third-party SIP device.




Detailed Instructions

1. In this subtask, you define a user called 1022 to associate with Line1 of the SPA112. The 1022 user name is configured with a password of 1234, a PIN of 1234, and Digest Credentials of 1234
 - a. Navigate to User Management > End User > Add New and complete the form appropriately based on your CUCM deployment:


Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

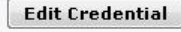

End User Configuration

 Save  Delete  Add New

Status

 Status: Ready

User Information

User Status	Active Local User
User ID*	<input type="text" value="1022"/>
Password	<input type="password" value="....."/> 
Confirm Password	<input type="password" value="....."/>
PIN	<input type="password" value="....."/> 
Confirm PIN	<input type="password" value="....."/>
Last name*	<input type="text" value="spa112 Line1"/>
Middle name	<input type="text"/>
First name	<input type="text" value="spa112 Line1"/>
Directory URI	<input type="text"/>
Telephone Number	<input type="text" value="1022"/>
Mail ID	<input type="text"/>
Manager User ID	<input type="text"/>
Department	<input type="text"/>
User Locale	<input type="text" value="English, United States"/>
Associated PC	<input type="text"/>
Digest Credentials	<input type="password" value="....."/>
Confirm Digest Credentials	<input type="password" value="....."/>

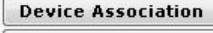
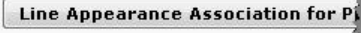
Service Settings

Home Cluster

Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)

UC Service Profile [View Details](#)

Device Information

Controlled Devices	<input type="text" value="SEPCCEF485C0E9B"/> 
Available Profiles	<input type="text" value="1001-EM
1002-EM
1003-EM"/> 

CTI Controlled Device Profiles

Extension Mobility

Available Profiles

Controlled Profiles

Default Profile

BLF Presence Group*

SUBSCRIBE Calling Search Space

Allow Control of Device from CTI

Enable Extension Mobility Cross Cluster

Directory Number Associations

Primary Extension

Mobility Information

Enable Mobility

Primary User Device

Enable Mobile Voice Access

Maximum Wait Time for Desk Pickup*

Remote Destination Limit*

Remote Destination Profiles

Multilevel Precedence and Preemption Authorization

MLPP User Identification Number

MLPP Password

MLPP Password	<input type="text"/>
Confirm MLPP Password	<input type="text"/>
MLPP Precedence Authorization Level	<input type="text" value="Routine"/>
CAPF Information	
Associated CAPF Profiles	<input type="text"/> View Details
Permissions Information	
Groups	<input type="text"/> View Details
Roles	<input type="text"/> View Details
<input type="button" value="Add to Access Control Group"/> <input type="button" value="Remove from Access Control Group"/>	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Add New"/>	

- b. Click Save
2. In this subtask, you will configure a basic third-party SIP device for Line1 of the SPA112 ATA as follows:
 - a. Navigate to Device > Phone > Add New > Phone Type > Third-party SIP Device (Basic) > Next
 - b. Insert the necessary device information appropriately based on your CUCM deployment:

Phone Type

Product Type: Third-party SIP Device (Basic)
Device Protocol: SIP

Device Information

Registration	Registered with Cisco Unified Communications Manager 10.99.31.140
IP Address	10.99.31.132
Active Load ID	Unknown
Download Status	Unknown
<input checked="" type="checkbox"/> Device is Active	
<input type="checkbox"/> Device is not trusted	
MAC Address*	CCEF485C0E9B
Description	CCEF485C0E9B spa112 line1
Device Pool*	PM_HQ_DP View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Third-party SIP Device (Basic)
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	PM_HQ_DEVICE_CSS
AAR Calling Search Space	< None >
Media Resource Group List	< None >
Location*	PM_HQ_LOC
AAR Group	< None >
Device Mobility Mode*	Off View Current Device Mobility Settings
Owner User ID	< None >
Use Trusted Relay Point*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Logged Into Hunt Group	
<input type="checkbox"/> Remote Device	

Call Routing Information

Inbound Calls

Calling Party Transformation CSS: < None >

Call Routing Information

Inbound Calls

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

Outbound Calls

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

Protocol Specific Information

BLF Presence Group* Standard Presence group

MTP Preferred Originating Codec* 711ulaw

Device Security Profile* Third-party SIP Device Basic - digest auth enabled

Rerouting Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard SIP Profile

Digest User 1022

Media Termination Point Required

Unattended Port

Require DTMF Reception

MLPP Information

MLPP Domain < None >

anfig Add New

- c. Click Save
3. In this subtask, you will associate a DN to the third-party SIP device that you have just defined.
 - a. Click Line [1] – Add a new DN

Phone Configuration Related Links: [Back To](#)

Save Delete Copy Reset Apply Config Add New

Status

i Status: Ready

Association Information

[Modify Button Items](#)


1 ■ ■ ■ [Line \[1\] - Add a new DN](#)

Phone Type

Product Type: Third-party SIP Device (Basic)
Device Protocol: SIP

Device Information

Registration: Unknown
IP Address: Unknown
 Device is Active
 Device is not trusted
MAC Address*: CCEF485C0E9B
Description: CCEF485C0E9B spa112 line1
Device Pool*: RM-HQ-CP



Note: you can also reach this link by navigating to Device > Phone > Find > Click the relevant Device Name > Line [1] – Add a new DN

- b. Insert the necessary device information for your CUCM deployment:

Directory Number Configuration
Related Links: [Configure Device](#)

Save Delete Reset Apply Config Add New

Status

Info Status: Ready

Directory Number Information

Directory Number*

Route Partition

Description

Alerting Name

ASCII Alerting Name

Associated Devices

▼ ▲

Dissociate Devices

Edit Device
Edit Line Appearance

Directory Number Settings

Voice Mail Profile (Choose <None> to use system default)

Calling Search Space

BLF Presence Group*

User Hold MOH Audio Source

Network Hold MOH Audio Source

Reject Anonymous Calls

Directory URIs

Primary	URI	Partition
<input checked="" type="radio"/>	<input type="text"/>	<input type="text" value="< None >"/>

Add Row

AAR Settings

AAR	Voice Mail	AAR Destination Mask	AAR O
<input type="checkbox"/>	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="< None >"/>

Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

Voice Mail	Destination	Calling Space
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>



AAR	<input type="checkbox"/> or	<input type="text"/>	< None >
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			
Call Forward and Call Pickup Settings			
	Voice Mail	Destination	Calling Search
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or	<input type="text"/>	< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Busy External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered External	<input type="checkbox"/> or	<input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group			< None >
Park Monitoring			
	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	< None > means to call the parker's line.
Park Monitoring Reversion Timer	<input type="text"/>		A blank value will use value set in Park Monitoring Reversion T
MLPP Alternate Party Settings			
Target (Destination)	<input type="text"/>		
MLPP Calling Search Space	< None >		
MLPP No Answer Ring Duration (seconds)	<input type="text"/>		
Line Settings for All Devices			
Hold Reversion Ring Duration (seconds)	<input type="text"/>		Setting the Hold Reversion Ring Duration to zero

Line Settings for All Devices

Hold Reversion Ring Duration (seconds) Setting the Hold Reversion Ring Duration to zero

Hold Reversion Notification Interval (seconds) Setting the Hold Reversion Notification Interval to

Party Entrance Tone*

Line 1 on Device SEPCCEF485C0E9B

Display (Caller ID) Display text for a line appearance is intended for displaying text such as a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Caller ID)

External Phone Number Mask

Monitoring Calling Search Space

Multiple Call/Call Waiting Settings on Device SEPCCEF485C0E9B

Note: The range to select the Max Number of calls is: 1-2

Maximum Number of Calls*

Busy Trigger* (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEPCCEF485C0E9B


Caller Name

Caller Number

Redirected Number

Dialed Number

Users Associated with Line

	Full Name	User ID	
<input type="checkbox"/>	spa112 Line1,spa112 Line1	1022	

- c. Associate the recently defined 1022 end user with the recently defined SPA122 Line 1 line. Click Associate End Users

Forwarded Call Information Display on Device SEPCCEF485C0E9B

Caller Name

Caller Number

Redirected Number

Dialed Number

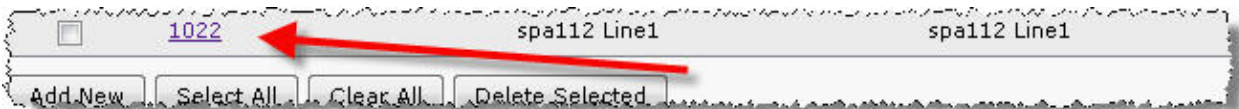
Users Associated with Line



- d. Select 1022 in the pop-up list



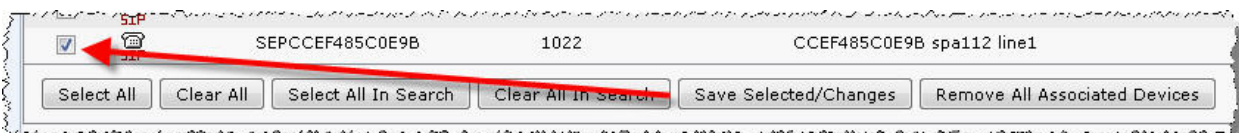
- e. Click Add Selected
 - f. Click Save
4. In this subtask, you will associate the SPA112 Line1 user with the SPA112 Line1 third-party SIP device.
- a. Navigate to User Management > End User > Click the 1022 link



- b. Navigate to Device Information > Device Association



- c. Select the SPA112 Line1 device



- d. Click Save Selected/Changes

This completes the minimum CUCM configuration required to support a SPA112 ATA.

Configuring the SPA112 ATA

Overview

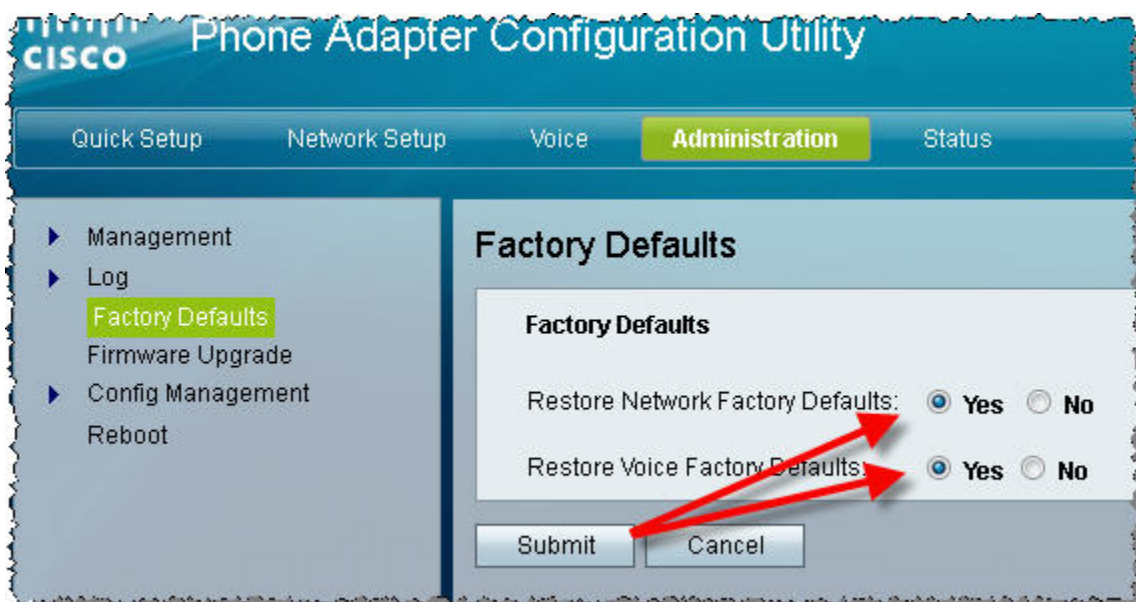
The following SPA112 administration tasks will be described in order to prepare the SPA112 for connecting the CUCM:

1. Defaulting the SPA112 to factory settings
2. Upgrading the SPA112 to the current firmware
3. Configuring Line 1 of the SPA112 to register to the CUCM

Detailed Instructions

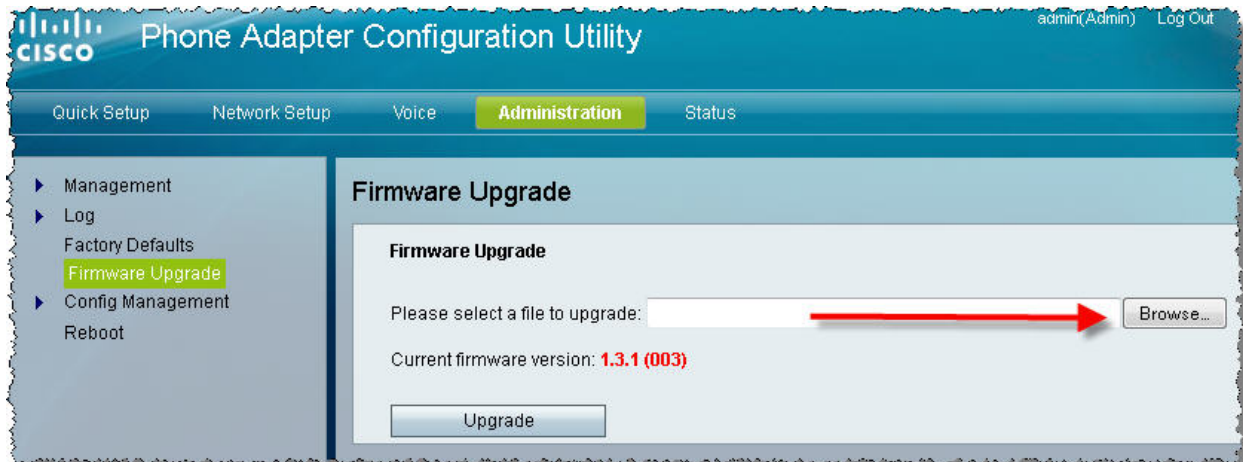
In this subtask, you restore the SPA112 to a factory default state so that you are starting from a known configuration.

1. Connect the SPA112 to your network. If you need help with connecting it and logging in, retrieve and follow the instructions in the most current version of the SPA112 Quick Start Guide located [here](#).
 - a. Access the web-User Interface (web-UI) at the **http://spa112IPaddress** URL.
 - b. Take note of the Version displayed under the "Phone Adapter Configuration Utility" text. If your SPA112 firmware is older than 1.2.1(004), you **must** use Internet Explorer until you have upgraded to at least 1.2.1(004) else you will not see some components of the web-UI
The default administration credentials are admin / admin
 - c. Navigate to Administration > Factory Defaults:
 - d. Select both Restore Network Factory Defaults and Restore Voice Factory Defaults



- e. Click Submit

2. In this subtask, you upgrade your SPA112 to the most currently available SPA112 firmware available from [here](#)
 - a. If your SPA112 firmware is older than 1.2.1(004), you **must** use Internet Explorer until you have upgraded to at least 1.2.1(004) else you will not see some components of the web-UI



- b. Navigate to Administration > Firmware Upgrade > Browse > select the SPA112 firmware you just downloaded > Click Upgrade
3. In this subtask, you configure Line 1 of the SPA112 to register to the CUCM
 - a. Log in to the SPA112 as the admin user. Default password is admin
 - b. Navigate to Voice tab > Line 1 tab:
 - i. SIP Settings > SIP Transport: TCP
 - ii. Proxy and Registration: Proxy: 10.99.31.140 [CUCM IP address]
 - iii. Subscriber Information:
 1. User ID: 1022
 2. Password: 1234
 3. Use Auth ID: yes
 4. Auth ID: 1022
 - iv. Dial Plan:
 1. Dial Plan: (*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxS0|xxxxxxxxxxxxx.**1xxx**)
Addition in **bold red** to allow 4-digit dialing starting with 1

Line 1

General

Line Enable:

Streaming Audio Server (SAS)

SAS Enable:

SAS DLG Refresh Intvl:

SAS Inbound RTP Sink:

NAT Settings

NAT Mapping Enable:

NAT Keep Alive Enable:

NAT Keep Alive Msg:

NAT Keep Alive Dest:

Network Settings

SIP ToS/DiffServ Value:

SIP CoS Value: [0-7]

RTP ToS/DiffServ Value:

RTP CoS Value: [0-7]

Network Jitter Level:

Jitter Buffer Adjustment:

SIP Settings

SIP Transport:

SIP Port:

SIP 100REL Enable:

EXT SIP Port:

Auth Resync-Reboot:

SIP Proxy-Require:

SIP Remote-Party-ID:

SIP GUID:

SIP Debug Option:

RTP Log Intvl:

Restrict Source IP:

Referor Bye Delay:

Refer Target Bye Delay:

Referee Bye Delay:

Refer-To Target Contact:

Sticky 183:

Auth INVITE:

Reply 182 On Call Waiting:

Use Anonymous With RPID:

Use Local Addr In FROM:

Call Feature Settings

Blind Attn-Xfer Enable:

MOH Server:

Xfer When Hangup Conf:

Conference Bridge URL:

Conference Bridge Ports:

Enable IP Dialing:

Emergency Number:

Mailbox ID:

Proxy and Registration

Proxy:

Outbound Proxy:

Use Outbound Proxy:

Use OB Proxy In Dialog:

Register:

Make Call Without Reg:

Register Expires:

Ans Call Without Reg:



Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal
Mailbox Subscribe URL:		Mailbox Subscribe Expires:	2147483647

Subscriber Information

Display Name:		User ID:	1022
Password:	*****	Use Auth ID:	yes
Auth ID:	1022	Resident Online Number:	
SIP URI:			

Supplementary Service Subscription

Call Waiting Serv:	yes	Block CID Serv:	yes
Block ANC Serv:	yes	Dist Ring Serv:	yes
Cfwd All Serv:	yes	Cfwd Busy Serv:	yes
Cfwd No Ans Serv:	yes	Cfwd Sel Serv:	yes
Cfwd Last Serv:	yes	Block Last Serv:	yes
Accept Last Serv:	yes	DND Serv:	yes
CID Serv:	yes	CWCID Serv:	yes
Call Return Serv:	yes	Call Redial Serv:	yes
Call Back Serv:	yes	Three Way Call Serv:	yes
Three Way Conf Serv:	yes	Attn Transfer Serv:	yes
Unattn Transfer Serv:	yes	MVM Serv:	yes
VMM Serv:	yes	Speed Dial Serv:	yes
Secure Call Serv:	yes	Referral Serv:	yes
Feature Dial Serv:	yes	Service Announcement Serv:	no
Reuse CID Number As Name:	yes		

Audio Configuration

Preferred Codec:	G711u	Second Preferred Codec:	Unspecified
Third Preferred Codec:	Unspecified	Use Pref Codec Only:	no
Use Remote Pref Codec:	no	Codec Negotiation:	Default
G729a Enable:	yes	Silence Supp Enable:	no
G726-32 Enable:	yes	Silence Threshold:	medium
FAX V21 Detect Enable:	yes	Echo Canc Enable:	yes
FAX CNG Detect Enable:	yes	FAX Passthru Codec:	G711u
FAX Codec Symmetric:	yes	DTMF Process INFO:	yes
FAX Passthru Method:	NSE	DTMF Process AVT:	yes

FAX Passthru Method:	NSE	DTMF Process AVT:	yes
FAX Process NSE:	yes	DTMF Tx Method:	Auto
FAX Disable ECAN:	no	DTMF Tx Mode:	Strict
DTMF Tx Strict Hold Off Time:	70	FAX Enable T38:	no
Hook Flash Tx Method:	None	FAX T38 Redundancy:	1
FAX T38 ECM Enable:	yes	FAX Tone Detect Mode:	caller or callee
Symmetric RTP:	no	FAX T38 Return to Voice:	no
Modem Line:	no		

Dial Plan

Dial Plan: (*xx[[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxxx|1xxx)

FXS Port Polarity Configuration

Idle Polarity: Forward Caller Conn Polarity: Forward

Callee Conn Polarity: Forward

Submit Cancel Refresh

c. Click Submit

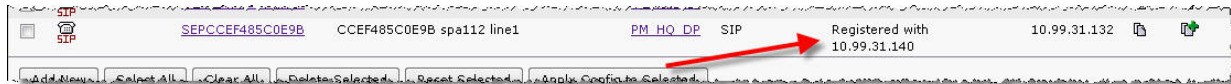
This completes the minimum SPA112 configuration required to register to a CUCM.

Verifying Registration Status

Knowing the registration status of your SPA112 is helpful when troubleshooting.

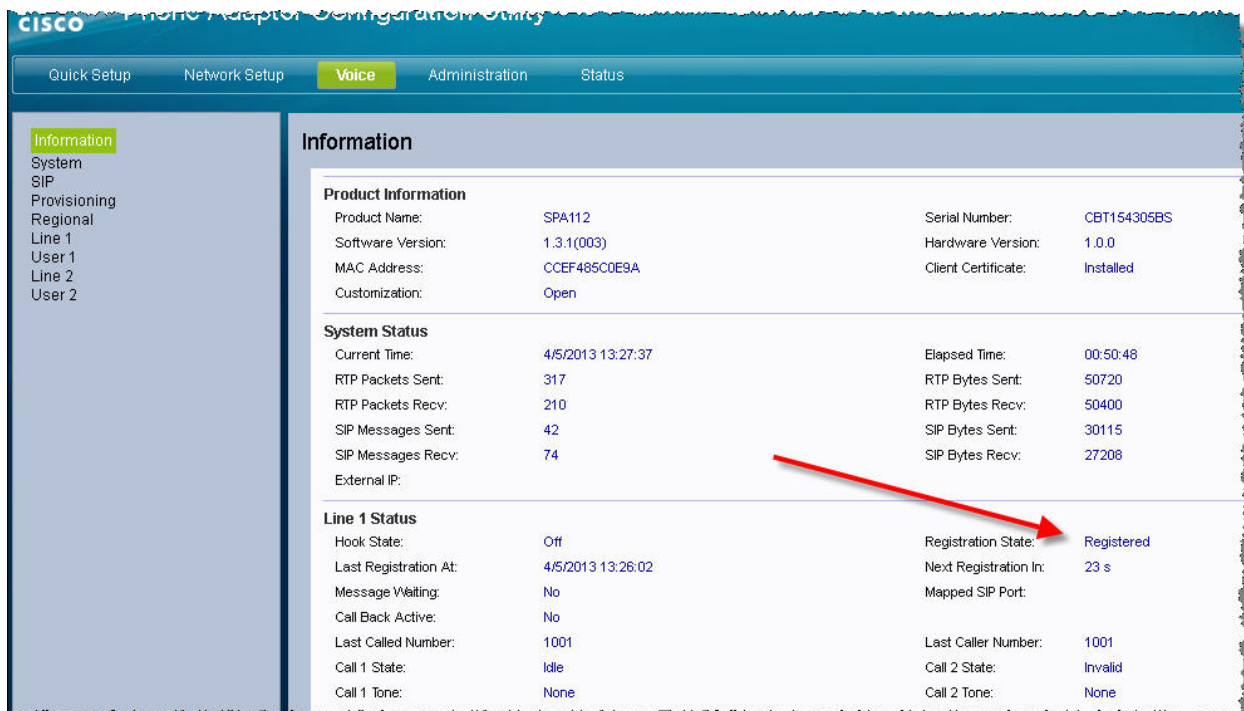
CUCM

View the SPA112 Line 1's registration status from the CUCM's point of view by navigating to Device > Phone > Find:



SPA112

View the SPA112 Line 1's registration status from the SPA112's point of view by navigating to the Voice tab > Information tab:





Testing

Once you have completed configuring the CUCM, the SPA112, and have verified that the SPA112's Line 1 is registered with the CUCM, you should test both inbound and outbound calls.

SPA112 Inbound Calls

From a known working phone, dial the extension associated with the Line 1 device of the SPA112, 1022 in this document. The analog phone connected to the SPA112 should ring. When you answer, verify that there is two-way audio.

SPA112 Outbound Calls

From the analog phone connected to the SPA112 PHONE 1 port, dial the extension of a known working phone registered to the CUCM, 1001 in this document. The target phone should ring. When you answer, verify that there is two-way audio.

SIP Registration Packet Trace

Following is a Wireshark packet capture of a successful SIP registration with the CUCM:

No.	Time	ToD	Source	Destination	Protocol	Length
325	69.454	14:36:55.516	10.99.31.132	10.99.31.140	SIP	647

Request: REGISTER sip:10.99.31.140 |

Frame 325: 647 bytes on wire (5176 bits), 647 bytes captured (5176 bits) on interface 0 Ethernet II, Src: Cisco_5c:0e:9a (cc:ef:48:5c:0e:9a), Dst: WistronI_11:22:33 (f0:de:f1:11:22:33)
Internet Protocol Version 4, Src: 10.99.31.132 (10.99.31.132), Dst: 10.99.31.140 (10.99.31.140)
Transmission Control Protocol, Src Port: 5077 (5077), Dst Port: sip (5060), Seq: 1, Ack: 1, Len: 581

Session Initiation Protocol (REGISTER)

Request-Line: REGISTER sip:10.99.31.140 SIP/2.0

Message Header

Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-564954c4

From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0

To: <sip:1022@10.99.31.140>

Call-ID: f81e24aa-db5053bf@10.99.31.132

CSeq: 26543 REGISTER

Max-Forwards: 70

Contact: <sip:1022@10.99.31.132:5077;transport=tcp>;expires=3600

P-Station-Name: ;mac=ccef485c0e9a; sn=CBT154305BS

User-Agent: Cisco/SPA112-1.3.1(003)

P-Station-Name: ;mac=ccef485c0e9a; display=""; sn=CBT154305BS

Content-Length: 0

Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER

Supported: replaces

No.	Time	ToD	Source	Destination	Protocol	Length
327	69.456	14:36:55.518	10.99.31.140	10.99.31.132	SIP	349

Status: 100 Trying (0 bindings) |

Frame 327: 349 bytes on wire (2792 bits), 349 bytes captured (2792 bits) on interface 0 Ethernet II, Src: WistronI_11:22:33 (f0:de:f1:11:22:33), Dst: Cisco_5c:0e:9a (cc:ef:48:5c:0e:9a)



Internet Protocol Version 4, Src: 10.99.31.140 (10.99.31.140), Dst: 10.99.31.132 (10.99.31.132)
Transmission Control Protocol, Src Port: sip (5060), Dst Port: 5077 (5077), Seq: 1, Ack: 582, Len: 283
Session Initiation Protocol (100)
Status-Line: SIP/2.0 100 Trying
Message Header
Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-564954c4
From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0
To: <sip:1022@10.99.31.140>
Date: Fri, 05 Apr 2013 19:36:51 GMT
Call-ID: f81e24aa-db5053bf@10.99.31.132
CSeq: 26543 REGISTER
Content-Length: 0

No.	Time	ToD	Source	Destination	Protocol	Length
328	69.457	14:36:55.519	10.99.31.140	10.99.31.132	SIP	472

Info
Status: 401 Unauthorized (0 bindings) |

Frame 328: 472 bytes on wire (3776 bits), 472 bytes captured (3776 bits) on interface 0 Ethernet II, Src: WistronI_11:22:33 (f0:de:f1:11:22:33), Dst: Cisco_5c:0e:9a (cc:ef:48:5c:0e:9a)
Internet Protocol Version 4, Src: 10.99.31.140 (10.99.31.140), Dst: 10.99.31.132 (10.99.31.132)
Transmission Control Protocol, Src Port: sip (5060), Dst Port: 5077 (5077), Seq: 284, Ack: 582, Len: 406
Session Initiation Protocol (401)
Status-Line: SIP/2.0 401 Unauthorized
Message Header
Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-564954c4
From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0
To: <sip:1022@10.99.31.140>;tag=2075367907
Date: Fri, 05 Apr 2013 19:36:51 GMT
Call-ID: f81e24aa-db5053bf@10.99.31.132
CSeq: 26543 REGISTER
WWW-Authenticate: Digest realm="ccmsipline",
nonce="iVST/8LeNp0TaD2e7s00mp9kyP7k7/w9", algorithm=MD5
Content-Length: 0

No.	Time	ToD	Source	Destination	Protocol	Length
329	69.458	14:36:55.520	10.99.31.132	10.99.31.140	SIP	647

Info
Request: REGISTER sip:10.99.31.140 |

Frame 329: 647 bytes on wire (5176 bits), 647 bytes captured (5176 bits) on interface 0 Ethernet II, Src: Cisco_5c:0e:9a (cc:ef:48:5c:0e:9a), Dst: WistronI_11:22:33 (f0:de:f1:11:22:33)
Internet Protocol Version 4, Src: 10.99.31.132 (10.99.31.132), Dst: 10.99.31.140 (10.99.31.140)
Transmission Control Protocol, Src Port: 5077 (5077), Dst Port: sip (5060), Seq: 582, Ack: 1, Len: 581
Session Initiation Protocol (REGISTER)
Request-Line: REGISTER sip:10.99.31.140 SIP/2.0
Message Header
Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-564954c4
From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0
To: <sip:1022@10.99.31.140>
Call-ID: f81e24aa-db5053bf@10.99.31.132
CSeq: 26543 REGISTER
Max-Forwards: 70
Contact: <sip:1022@10.99.31.132:5077;transport=tcp>;expires=3600
P-Station-Name: ;mac=ccef485c0e9a; sn=CBT154305BS
User-Agent: Cisco/SPA112-1.3.1(003)



P-Station-Name: ;mac=ccef485c0e9a; display=""; sn=CBT154305BS
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: replaces

No.	Time	ToD	Source	Destination	Protocol	Length
332	69.459	14:36:55.521	10.99.31.140	10.99.31.132	SIP	349

Info
Status: 100 Trying (0 bindings) |

Frame 332: 349 bytes on wire (2792 bits), 349 bytes captured (2792 bits) on interface 0
Ethernet II, Src: WistronI_11:22:33 (f0:de:f1:11:22:33), Dst: Cisco_5c:0e:9a (cc:ef:48:5c:0e:9a)
Internet Protocol Version 4, Src: 10.99.31.140 (10.99.31.140), Dst: 10.99.31.132 (10.99.31.132)
Transmission Control Protocol, Src Port: sip (5060), Dst Port: 5077 (5077), Seq: 690, Ack: 1163, Len: 283
Session Initiation Protocol (100)
Status-Line: SIP/2.0 100 Trying
Message Header
Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-564954c4
From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0
To: <sip:1022@10.99.31.140>
Date: Fri, 05 Apr 2013 19:36:51 GMT
Call-ID: f81e24aa-db5053bf@10.99.31.132
CSeq: 26543 REGISTER
Content-Length: 0

No.	Time	ToD	Source	Destination	Protocol	Length
333	69.460	14:36:55.522	10.99.31.140	10.99.31.132	SIP	472

Info
Status: 401 Unauthorized (0 bindings) |

Frame 333: 472 bytes on wire (3776 bits), 472 bytes captured (3776 bits) on interface 0
Ethernet II, Src: WistronI_11:22:33 (f0:de:f1:11:22:33), Dst: Cisco_5c:0e:9a (cc:ef:48:5c:0e:9a)
Internet Protocol Version 4, Src: 10.99.31.140 (10.99.31.140), Dst: 10.99.31.132 (10.99.31.132)
Transmission Control Protocol, Src Port: sip (5060), Dst Port: 5077 (5077), Seq: 973, Ack: 1163, Len: 406
Session Initiation Protocol (401)
Status-Line: SIP/2.0 401 Unauthorized
Message Header
Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-564954c4
From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0
To: <sip:1022@10.99.31.140>;tag=1680687713
Date: Fri, 05 Apr 2013 19:36:51 GMT
Call-ID: f81e24aa-db5053bf@10.99.31.132
CSeq: 26543 REGISTER
WWW-Authenticate: Digest realm="ccmsipline",
nonce="iVST/8LeNp0Tad2e7s00mp9kyP7k7/w9", algorithm=MD5
Content-Length: 0

No.	Time	ToD	Source	Destination	Protocol	Length
336	69.472	14:36:55.534	10.99.31.132	10.99.31.140	SIP	827

Info
Request: REGISTER sip:10.99.31.140 |

Frame 336: 827 bytes on wire (6616 bits), 827 bytes captured (6616 bits) on interface 0
Ethernet II, Src: Cisco_5c:0e:9a (cc:ef:48:5c:0e:9a), Dst: WistronI_11:22:33 (f0:de:f1:11:22:33)
Internet Protocol Version 4, Src: 10.99.31.132 (10.99.31.132), Dst: 10.99.31.140 (10.99.31.140)



Transmission Control Protocol, Src Port: 5077 (5077), Dst Port: sip (5060), Seq: 1163, Ack: 1379, Len: 761

Session Initiation Protocol (REGISTER)

Request-Line: REGISTER sip:10.99.31.140 SIP/2.0

Message Header

Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-c6ef563c

From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0

To: <sip:1022@10.99.31.140>

Call-ID: f81e24aa-db5053bf@10.99.31.132

CSeq: 26544 REGISTER

Max-Forwards: 70

Authorization: Digest

username="1022",realm="ccmsipline",nonce="ivST/8LeNp0TaD2e7s00mp9kyP7k7/w9",uri="sip:10.99.31.140",algorithm=MD5,response="003c24b4c124aa7f5019dd5a73a71346"

Contact: <sip:1022@10.99.31.132:5077;transport=tcp>;expires=3600

P-Station-Name: ;mac=ccef485c0e9a; sn=CBT154305BS

User-Agent: Cisco/SPA112-1.3.1(003)

P-Station-Name: ;mac=ccef485c0e9a; display=""; sn=CBT154305BS

Content-Length: 0

Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER

Supported: replaces

No.	Time	ToD	Source	Destination	Protocol	Length
337	69.488	14:36:55.550	10.99.31.140	10.99.31.132	SIP	349

Status: 100 Trying (0 bindings) |

Frame 337: 349 bytes on wire (2792 bits), 349 bytes captured (2792 bits) on interface 0

Ethernet II, Src: WistronI_11:22:33 (f0:de:f1:11:22:33), Dst: Cisco_5c:0e:9a

(cc:ef:48:5c:0e:9a)

Internet Protocol Version 4, Src: 10.99.31.140 (10.99.31.140), Dst: 10.99.31.132

(10.99.31.132)

Transmission Control Protocol, Src Port: sip (5060), Dst Port: 5077 (5077), Seq: 1379, Ack: 1924, Len: 283

Session Initiation Protocol (100)

Status-Line: SIP/2.0 100 Trying

Message Header

Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-c6ef563c

From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0

To: <sip:1022@10.99.31.140>

Date: Fri, 05 Apr 2013 19:36:51 GMT

Call-ID: f81e24aa-db5053bf@10.99.31.132

CSeq: 26544 REGISTER

Content-Length: 0

No.	Time	ToD	Source	Destination	Protocol	Length
338	69.498	14:36:55.560	10.99.31.140	10.99.31.132	SIP	471

Status: 200 OK (1 bindings) |

Frame 338: 471 bytes on wire (3768 bits), 471 bytes captured (3768 bits) on interface 0

Ethernet II, Src: WistronI_11:22:33 (f0:de:f1:11:22:33), Dst: Cisco_5c:0e:9a

(cc:ef:48:5c:0e:9a)

Internet Protocol Version 4, Src: 10.99.31.140 (10.99.31.140), Dst: 10.99.31.132

(10.99.31.132)

Transmission Control Protocol, Src Port: sip (5060), Dst Port: 5077 (5077), Seq: 1662, Ack: 1924, Len: 405

Session Initiation Protocol (200)

Status-Line: SIP/2.0 200 OK

Message Header

Via: SIP/2.0/TCP 10.99.31.132:5077;branch=z9hG4bK-c6ef563c

From: <sip:1022@10.99.31.140>;tag=212cfce5c8d03e9o0

To: <sip:1022@10.99.31.140>;tag=552965170

Date: Fri, 05 Apr 2013 19:36:51 GMT

Call-ID: f81e24aa-db5053bf@10.99.31.132
CSeq: 26544 REGISTER
Expires: 120
Contact: <sip:1022@10.99.31.132:5077;transport=tcp>;x-cisco-newreg
Supported: X-cisco-sis-6.0.0
Content-Length: 0

Troubleshooting:

Reboot SPA112 after making any changes on the CUCM to ensure that the SPA112 appropriately registers to the CUCM.

SIP REGISTER: 404: Not found:

Warning 399 "Unable to find device/user in database"

The SPA112 fails to register with the CUCM. Monitoring the network interaction and filtering for SIP, you see a SIP status message of 404 Not Found

```
214 09:54:14.342 10.99.31.140 10.99.31.132 SIP 435 Status: 404 Not Found (0 bindings) |
```

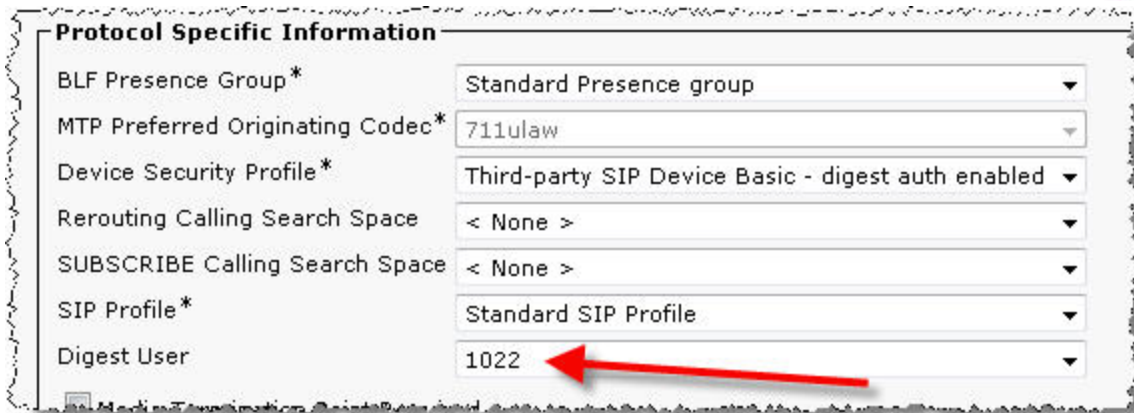
Looking at the SIP Message Header, you see:

```
Session initiated by 10.99.31.140 to 10.99.31.132 (404)
+ Status-Line: SIP/2.0 404 Not Found
+ Message Header
+ Via: SIP/2.0/UDP 10.99.31.132:5060;branch=z9hG4bK-3ed56c8
+ From: "spa112Line1" <sip:1021@10.99.31.140>;tag=9c5eb76ba51677d2e0
+ To: "spa112Line1" <sip:1021@10.99.31.140>;tag=1305189688
  Date: Thu, 04 Apr 2013 14:54:11 GMT
  Call-ID: a64e7dae-39622d6f@10.99.31.132
+ CSeq: 55460 REGISTER
  Warning: 399 cucmbe6k "Unable to find device/user in database"
  Content-Length: 0
```

Warning: 399 "Unable to find device/user in database"

Possible Solution

Define the 1022 user as the phone Digest User at Device > Phone > Find > Click the SPA112 Line 1 Link > Protocol Specific Information > Digest User: 1022

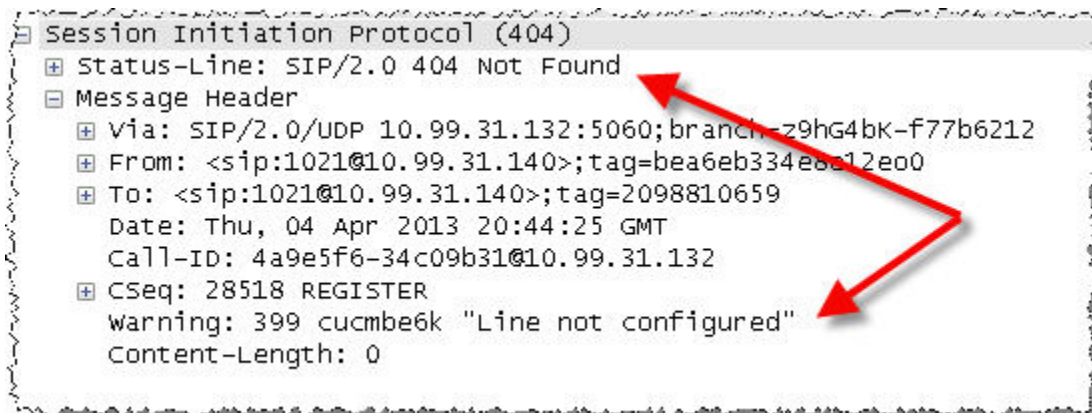


SIP REGISTER: 404: Not found: Warning 399 "Line not configured"

The SPA112 fails to register with the CUCM. Monitoring the network interaction and filtering for SIP, you see a SIP status message of 404 Not Found:

```
65647 15:42:28.693 10.99.31.140 10.99.31.132 SIP 387 Status: 404 Not Found (0 bindings)
```

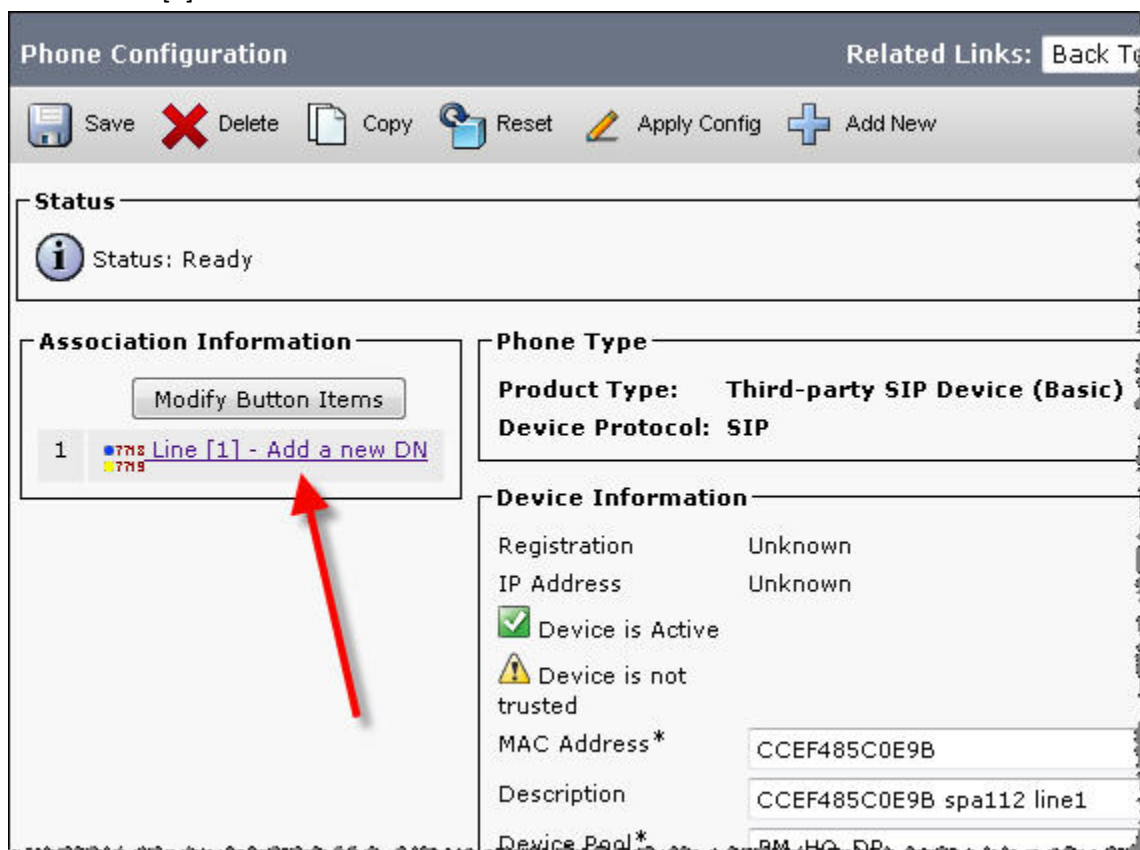
Looking at the SIP Message Header, you see:



Warning: 399 "Line not configured"

Possible Solution

Verify that a DN is assigned to Line [1] by navigating to Device > Phone > Find > Click the relevant Device Name > Line [1] – Add a new DN

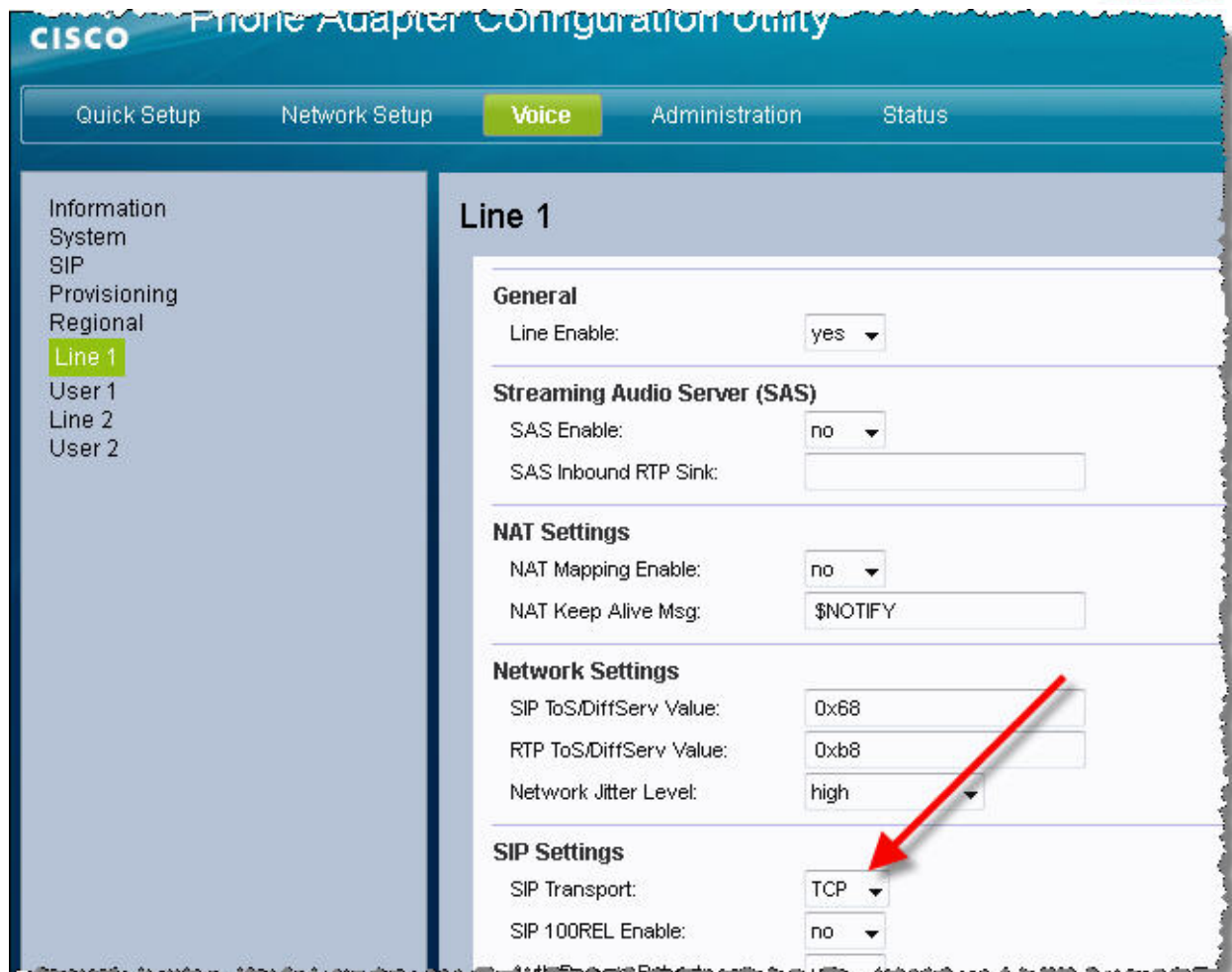


Outbound Call Fails with "Call cannot be completed as dialed" Message

The SPA112 is able to receive calls without problem but when attempting to make an outbound call, you hear a message saying "Call cannot be completed as dialed".

Possible Solution

Verify that the SPA112 is using the same transport protocol that the CUCM is expecting by default. By default, the SPA112 uses UDP as the SIP transport while CUCM expects TCP. Modify the SPA112 to use TCP as the transport by changing Voice tab > Line 1 tab > SIP Settings > SIP Transport: TCP



Additional Resources

- [SPA112 resources document](#) contains links to the quick start guide, user guide, admin guide, provisioning guide, firmware release notes, firmware versions, application notes, and troubleshooting tips
- [CUCM Documentation Guide](#) contains links to all relevant CUCM documents

<end>