Configuring the SPA112 ATA to operate with the CUCM

Contents

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.

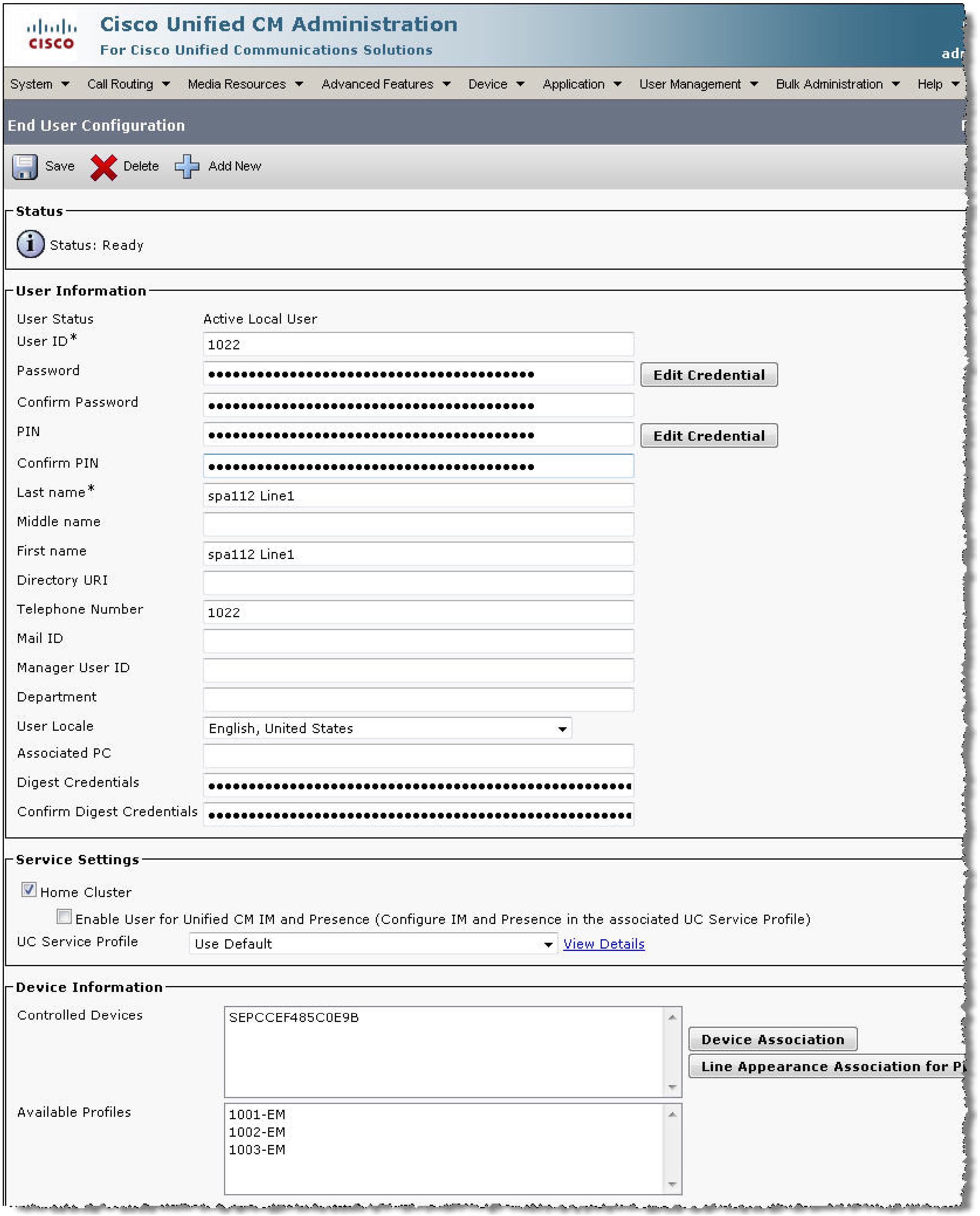
Overview

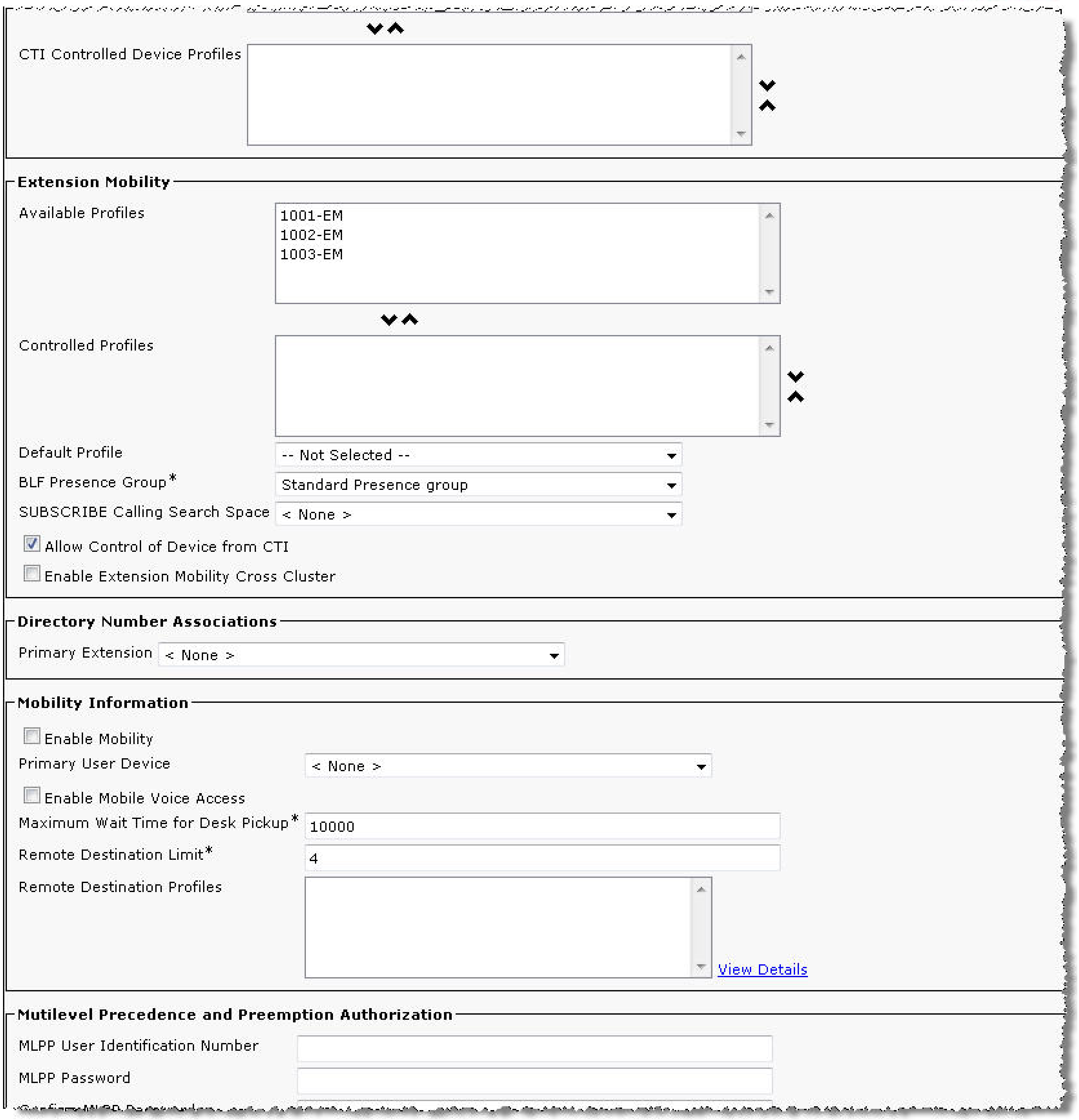
The following CUCM administration tasks to prepare the CUCM for connecting a SPA112 are described in detail:

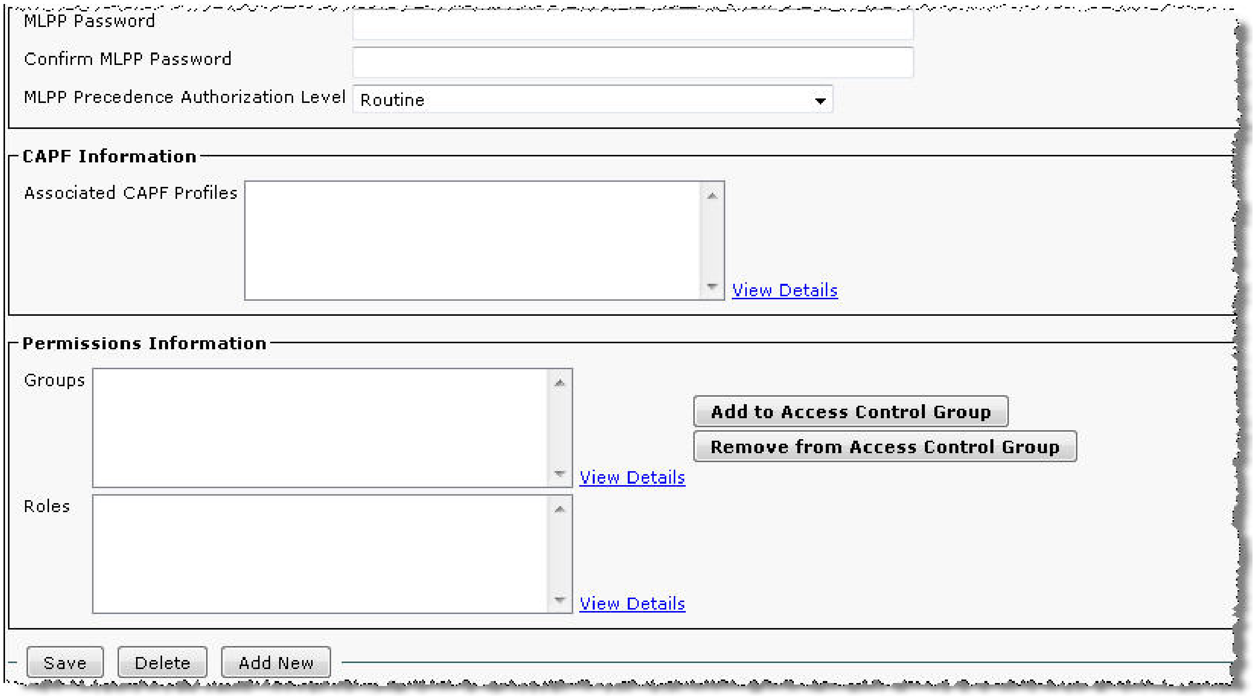
1. Define a CUCM user that will be associated with the SPA112 Line1
2. Define a third-party SIP phone for the SPA112 Line1
3. Associate a Line / DN (directory number) to the previously defined SPA112 Line1 third-party SIP device
4. Associate the SPA112 Line1 user with the SPA112 Line1 third-party SIP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

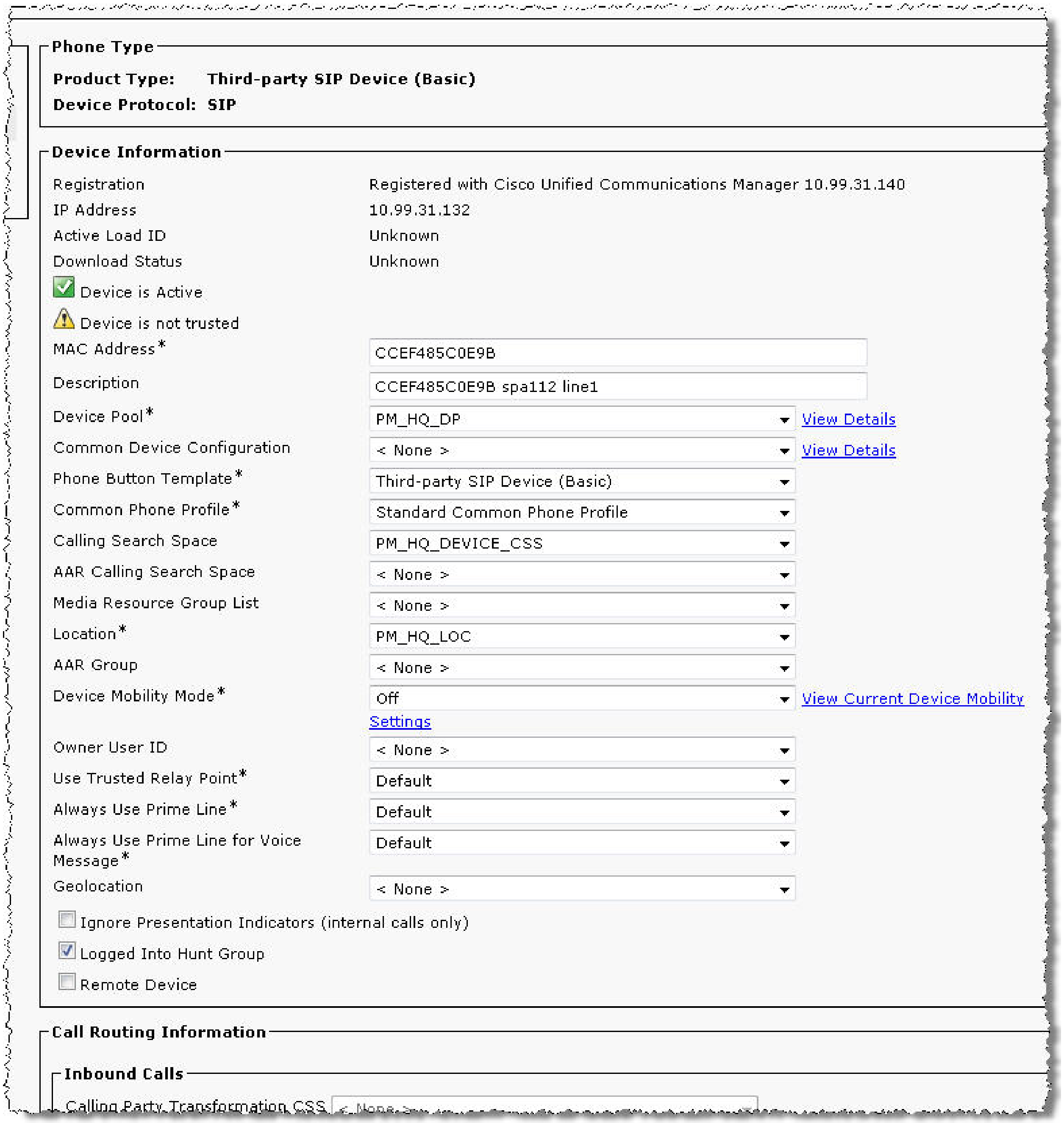

Cisco Unified CM Administration
For Cisco Unified Communications Solutions
System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
End User Configuration
Save
Delete
Add New
Status
Status: Ready
User Information
User Status
Active Local User
User ID*
1022
Password
Edit Credential
Confirm Password
PIN
Edit Credential
Confirm PIN
Last name*
spa112 Line1
Middle name
First name
spa112 Line1
Directory URI
Telephone Number
1022
Mail ID
Manager User ID
Department
User Locale
English, United States
Associated PC
Digest Credentials
Confirm Digest Credentials
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
Use Default
View Details
Device Information
Controlled Devices
SEPCCEF485C0E9B
Device Association
Available Profiles
1001-EM
1002-EM
1003-EM


CTI Controlled Device Profiles
Extension Mobility
Available Profiles
1001-EM
1002-EM
1003-EM
Controlled Profiles
Default Profile
-- Not Selected --
BLF Presence Group*
Standard Presence group
SUBSCRIBE Calling Search Space
< None >
Allow Control of Device from CTI
Enable Extension Mobility Cross Cluster
Directory Number Associations
Primary Extension
< None >
Mobility Information
Enable Mobility
Primary User Device
< None >
Enable Mobile Voice Access
Maximum Wait Time for Desk Pickup*
10000
Remote Destination Limit*
4
Remote Destination Profiles
View Details
Mutilevel Precedence and Preemption Authorization
MLPP User Identification Number
MLPP Password



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

Device is Active
Device is not trusted

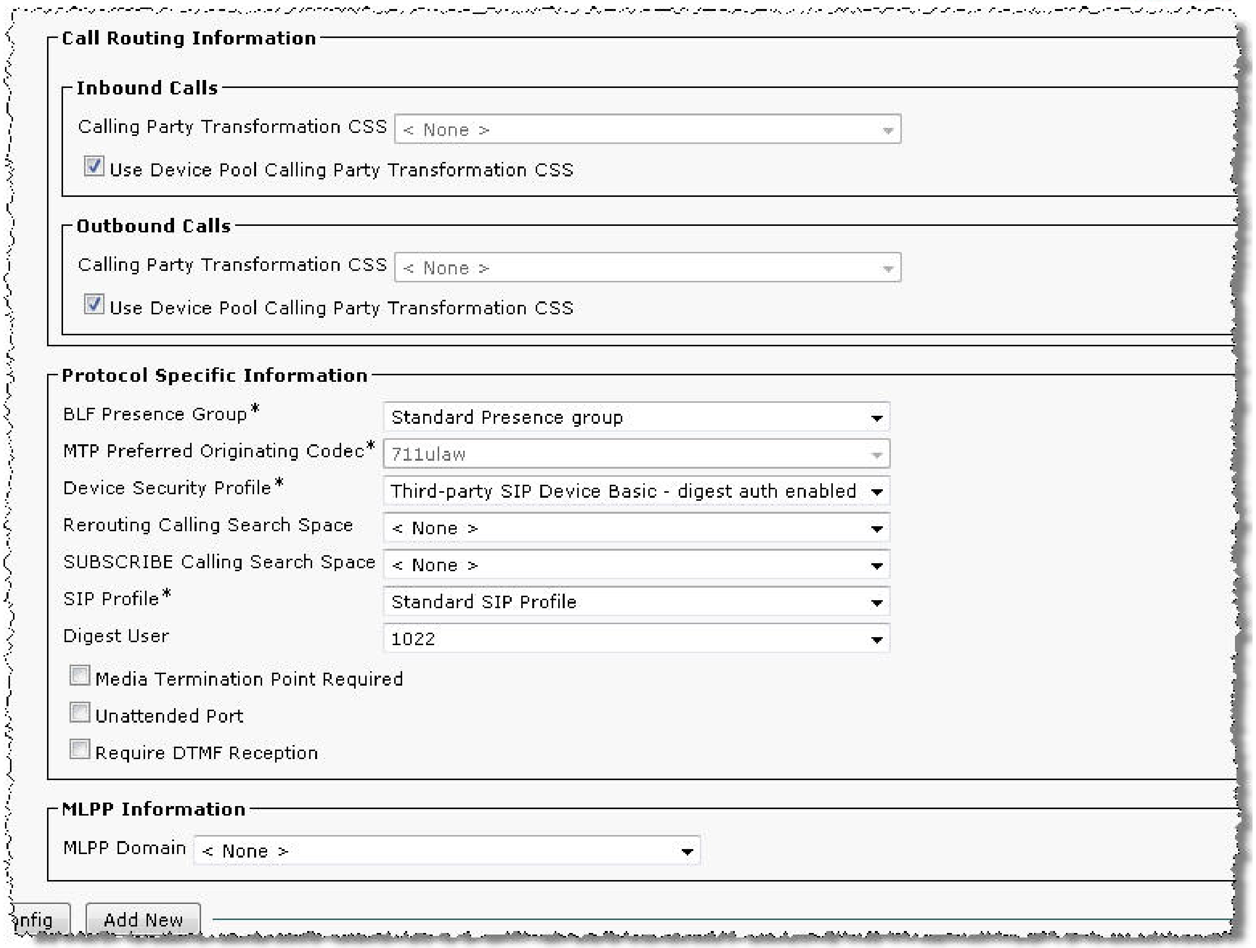
MAC Address*	CCEF485C0E9B	
Description	CCEF485C0E9B spa112 line1	
Device Pool*	PM_HQ_DP	View Details
Common Device Configuration	< None >	View Details
Phone Button Template*	Third-party SIP Device (Basic)	
Common Phone Profile*	Standard Common Phone Profile	
Calling Search Space	PM_HQ_DEVICE_CSS	
AAR 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 >	

- [ ] Ignore Presentation Indicators (internal calls only)
- [x] Logged Into Hunt Group
- [ ] Remote Device

Call Routing Information

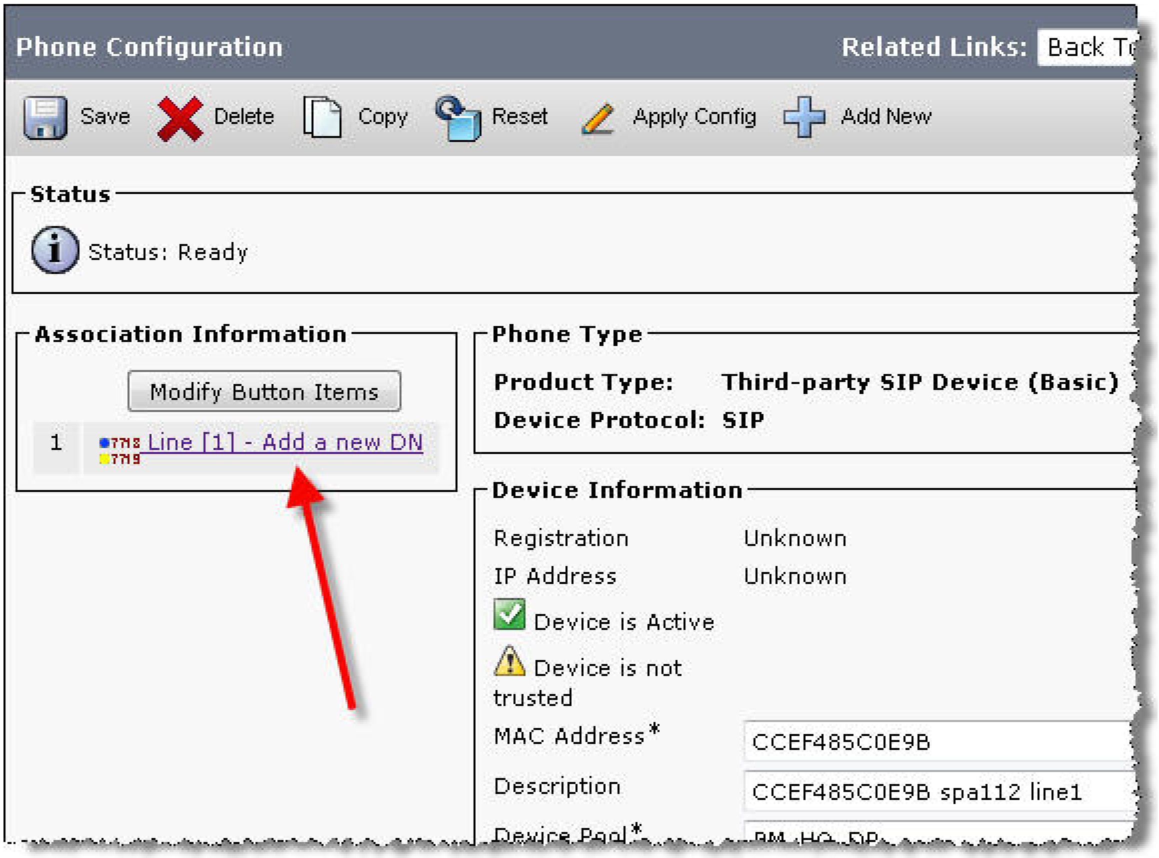
Inbound Calls

Calling Party Transformation CSS < None >



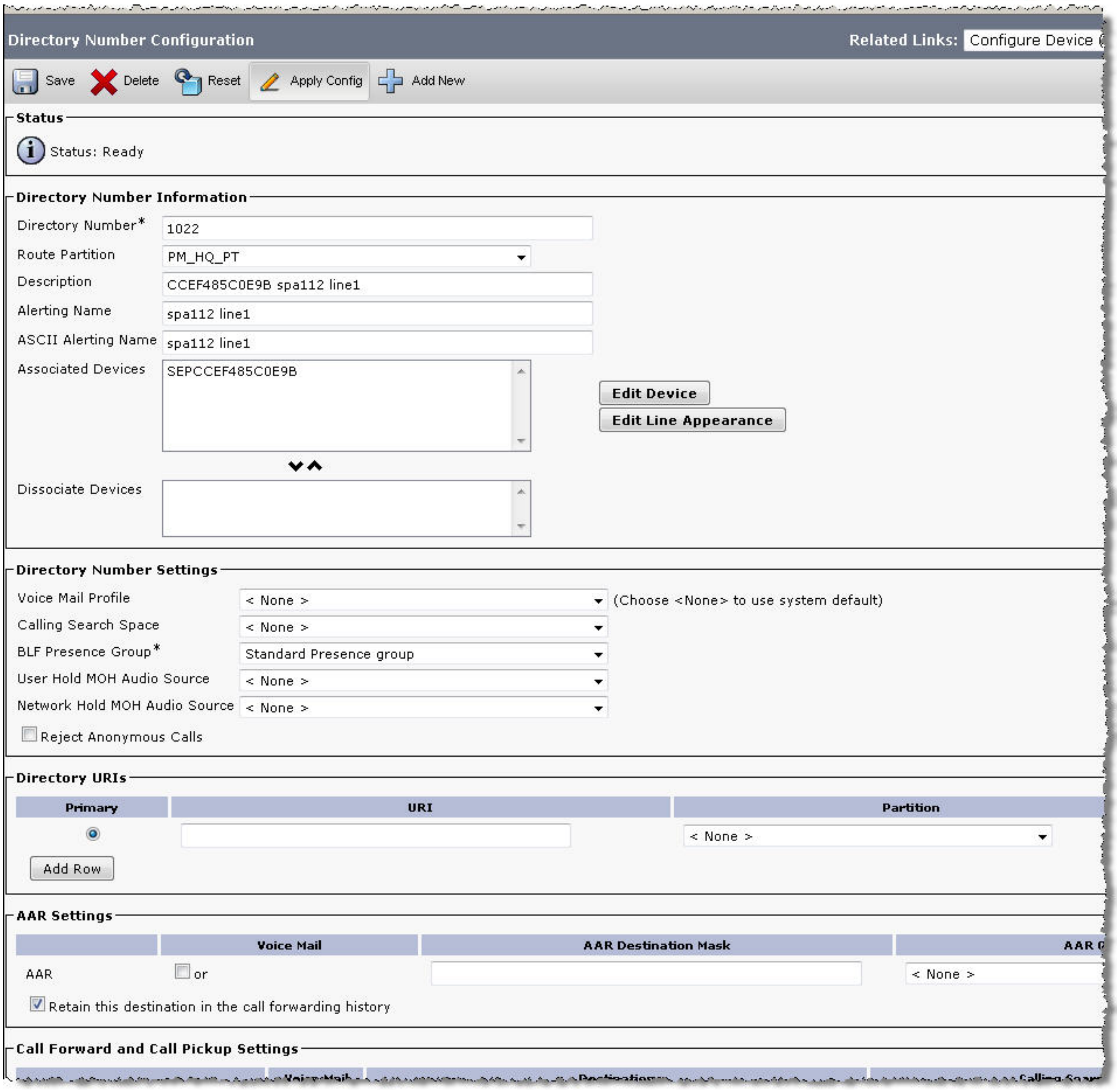
   c. Click Save

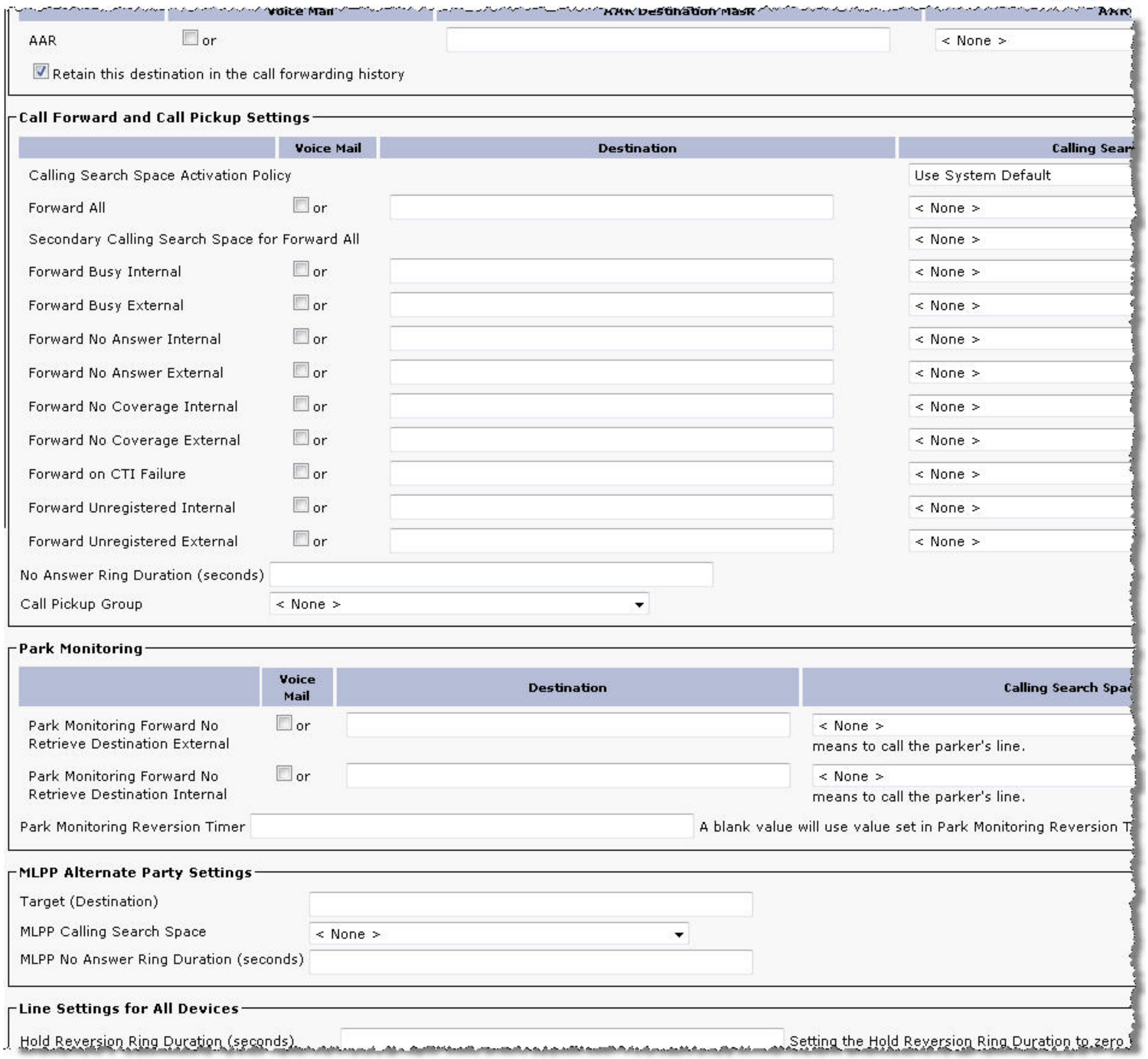
3. In this subtask, you will associate a DN to the third-party SIP device that you have just defined.
   a. Click Line [1} – Add a new DN



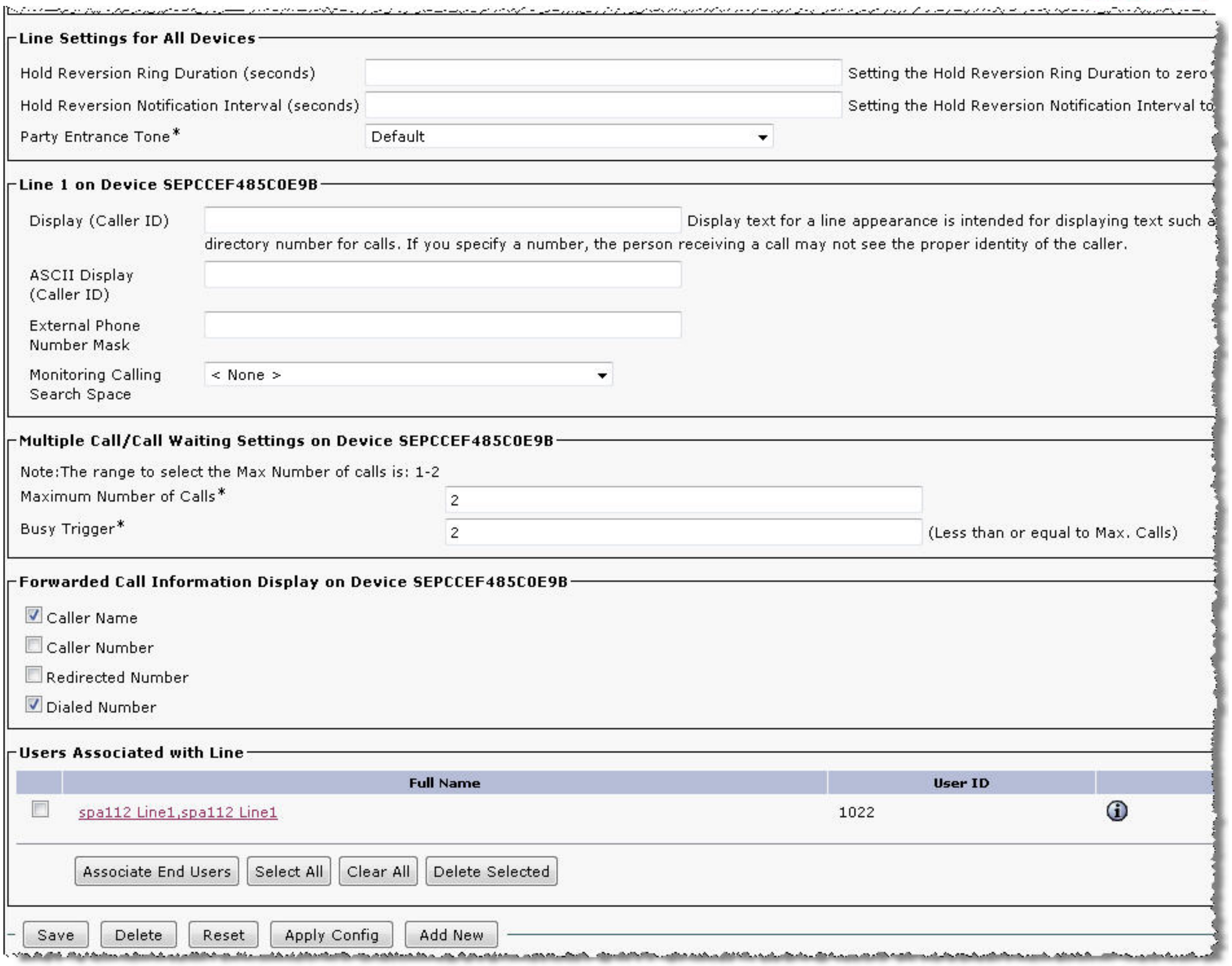
Note: you can also reach this link by navigating to Device > Phone > Find > Click the relevant Device Name > Line [1] – Add a new DN

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

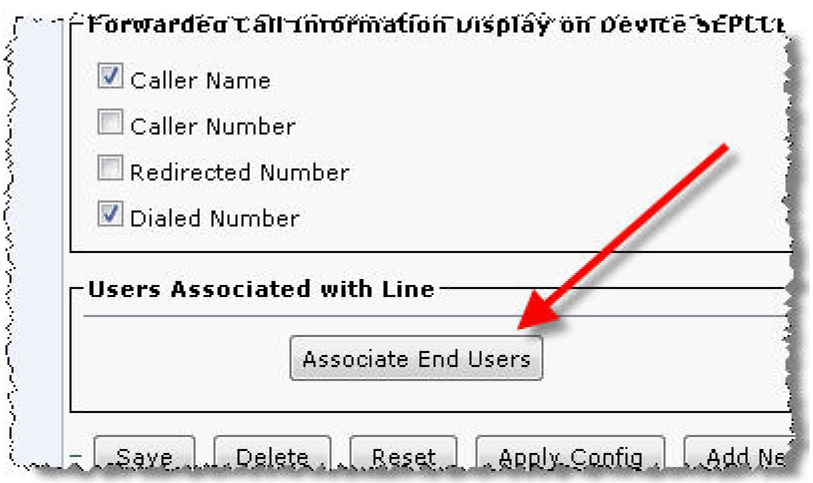

Directory Number Configuration
Related Links: Configure Device
Save
Delete
Reset
Apply Config
Add New
Status
Status: Ready
Directory Number Information
Directory Number*
1022
Route Partition
PM_HQ_PT
Description
CCEF485C0E9B spa112 line1
Alerting Name
spa112 line1
ASCII Alerting Name
spa112 line1
Associated Devices
SEPCCEF485C0E9B
Edit Device
Edit Line Appearance
Dissociate Devices
Directory Number Settings
Voice Mail Profile
< None >
(Choose <None> to use system default)
Calling Search Space
< None >
BLF Presence Group*
Standard Presence group
User Hold MOH Audio Source
< None >
Network Hold MOH Audio Source
< None >
Reject Anonymous Calls
Directory URIs
Primary
URI
Partition
< None >
Add Row
AAR Settings
Voice Mail
AAR Destination Mask
AAR
or
< None >
Retain this destination in the call forwarding history
Call Forward and Call Pickup Settings



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



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



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

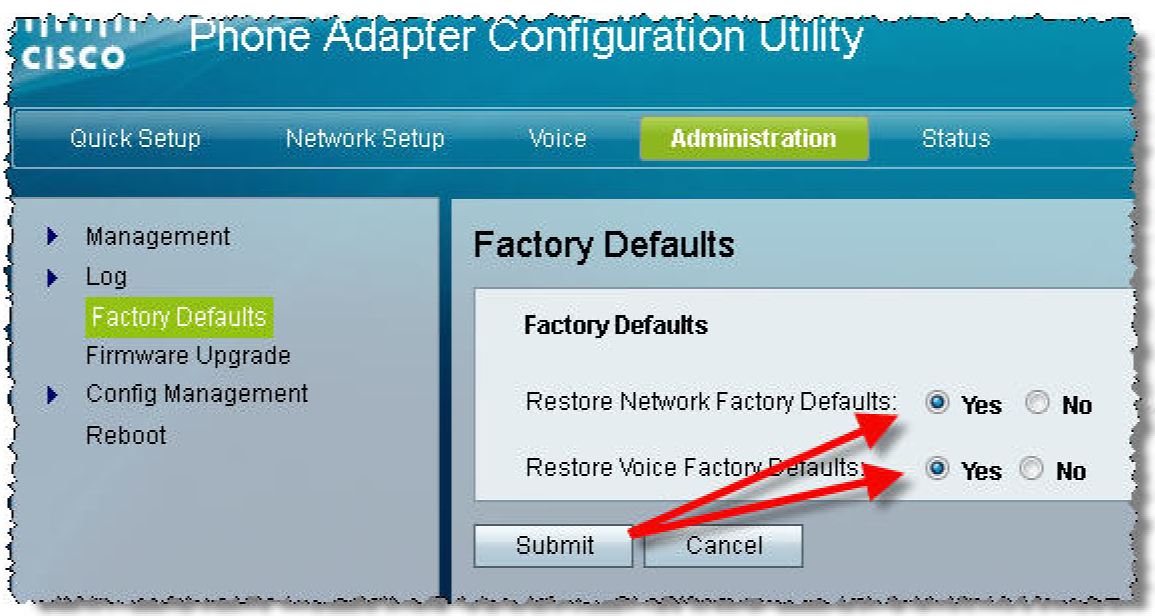
The following SPA112 administration tasks will be described in order to prepare the SPA112 for connecting the CUCM:

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

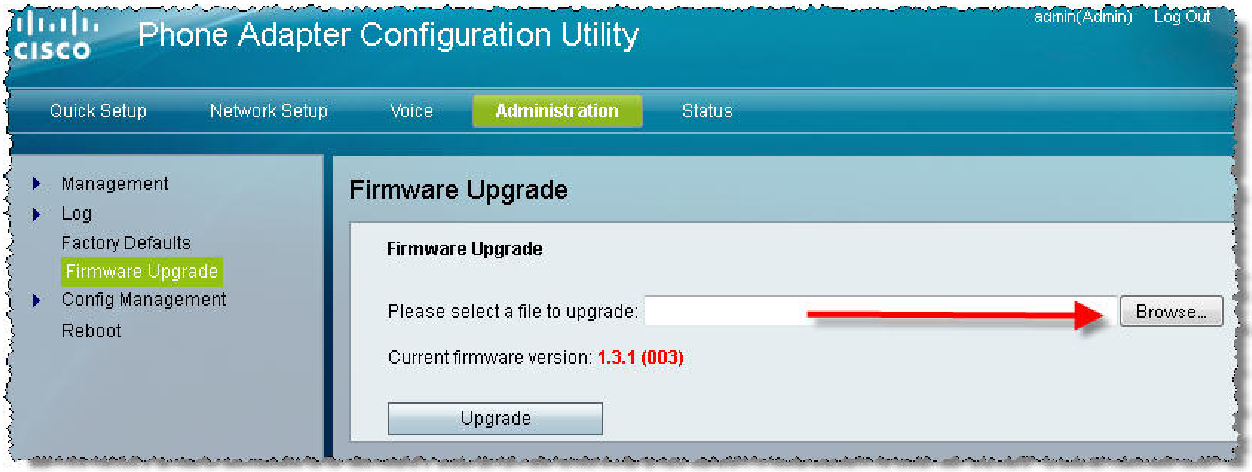
In this subtask, you restore the SPA112 to a factory default state so that you are starting from a known configuration.

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

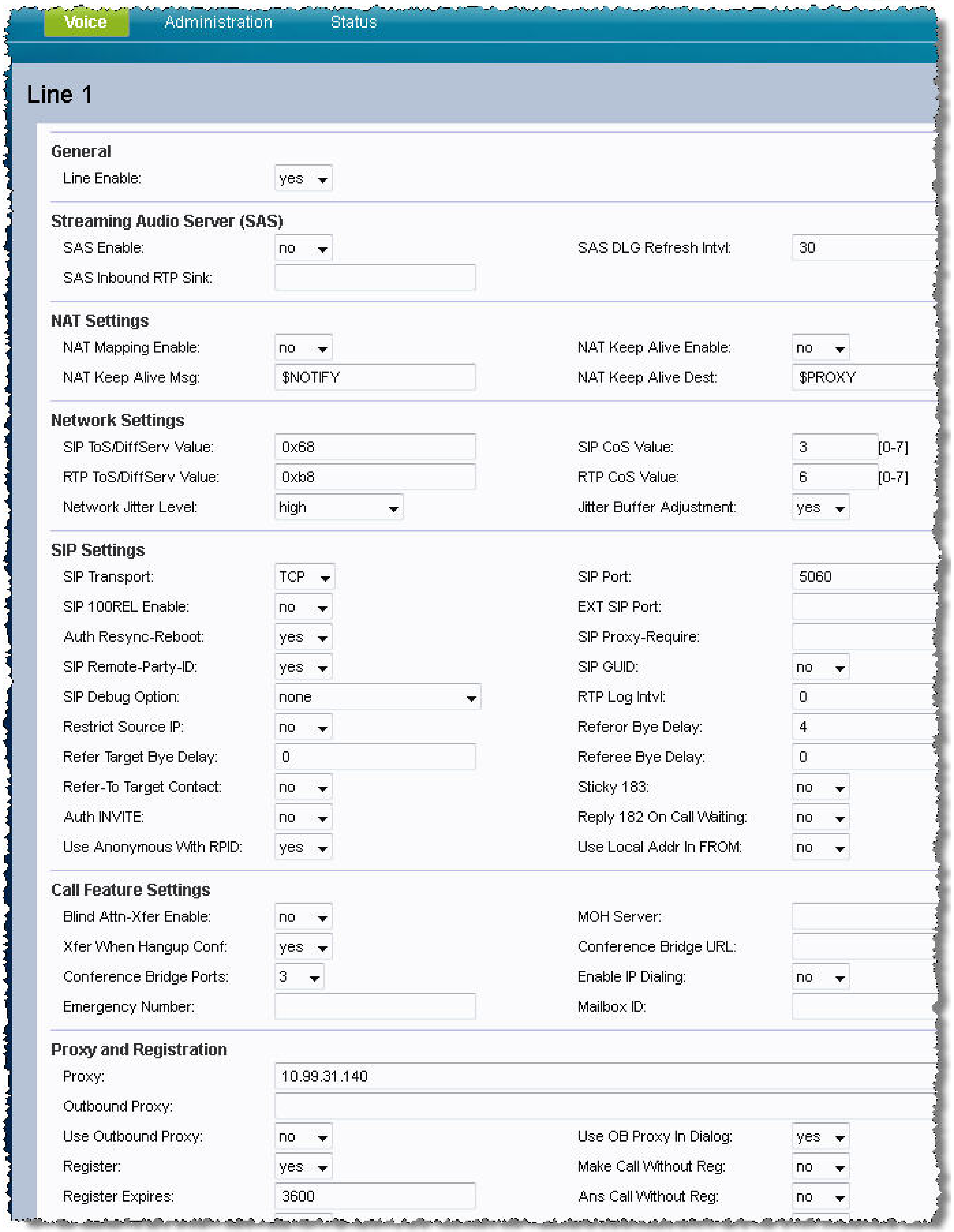


    e. Click Submit

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]xxxxxxS0|xxxxxxxxxxxx.**|1xxx**)
               Addition in **bold red** to allow 4-digit dialing starting with 1



Voice | Administration | Status

Line 1

General

Line Enable:	yes

Streaming Audio Server (SAS)

SAS Enable:	no	SAS DLG Refresh Intvl:	30
SAS Inbound RTP Sink:			

NAT Settings

NAT Mapping Enable:	no	NAT Keep Alive Enable:	no
NAT Keep Alive Msg:	$NOTIFY	NAT Keep Alive Dest:	$PROXY

Network Settings

SIP ToS/DiffServ Value:	0x68	SIP CoS Value:	3 [0-7]
RTP ToS/DiffServ Value:	0xb8	RTP CoS Value:	6 [0-7]
Network Jitter Level:	high	Jitter Buffer Adjustment:	yes

SIP Settings

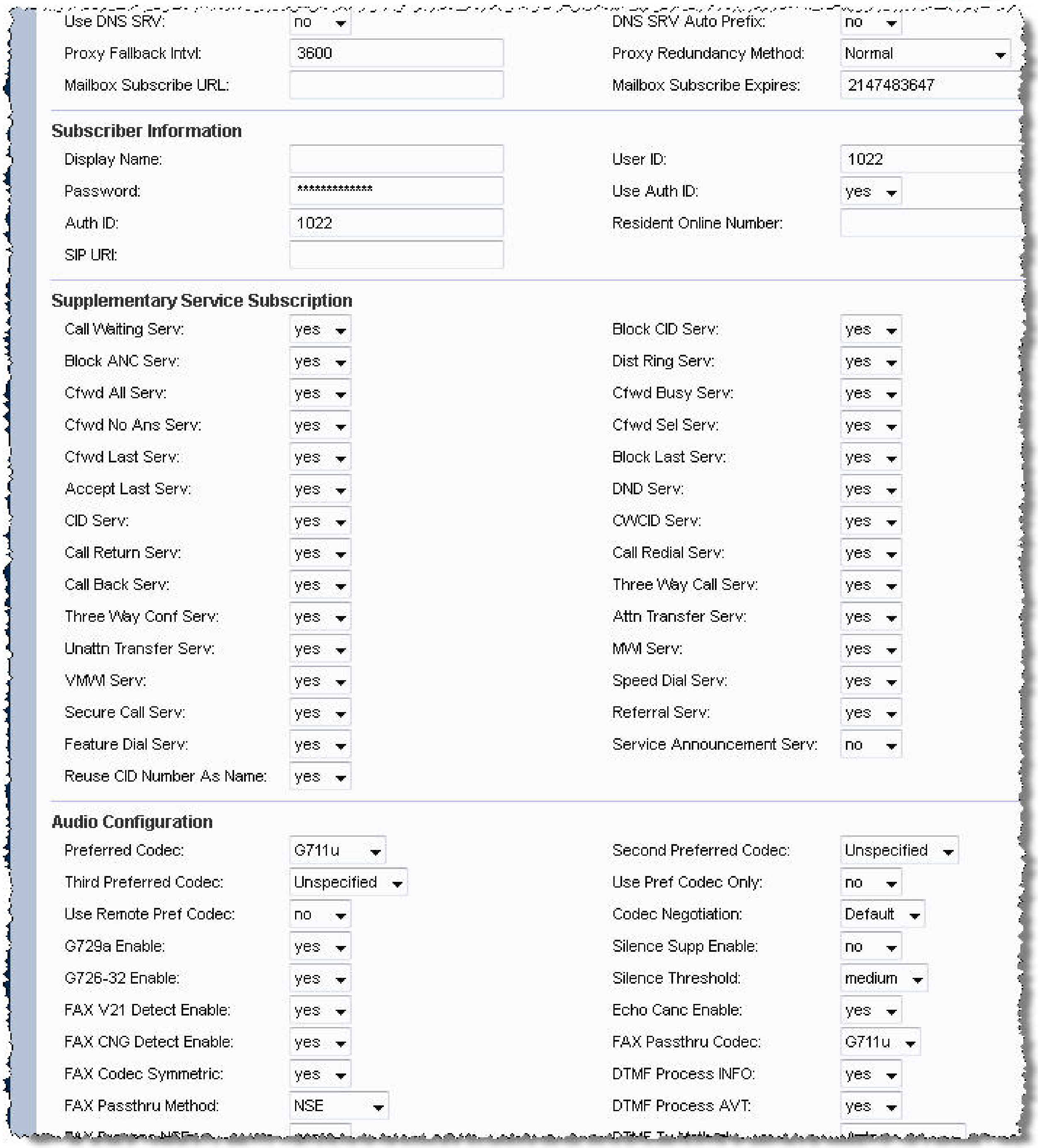
SIP Transport:	TCP	SIP Port:	5060
SIP 100REL Enable:	no	EXT SIP Port:	
Auth Resync-Reboot:	yes	SIP Proxy-Require:	
SIP Remote-Party-ID:	yes	SIP GUID:	no
SIP Debug Option:	none	RTP Log Intvl:	0
Restrict Source IP:	no	Referor Bye Delay:	4
Refer Target Bye Delay:	0	Referee Bye Delay:	0
Refer-To Target Contact:	no	Sticky 183:	no
Auth INVITE:	no	Reply 182 On Call Waiting:	no
Use Anonymous With RPID:	yes	Use Local Addr In FROM:	no

Call Feature Settings

Blind Attn-Xfer Enable:	no	MOH Server:	
Xfer When Hangup Conf:	yes	Conference Bridge URL:	
Conference Bridge Ports:	3	Enable IP Dialing:	no
Emergency Number:		Mailbox ID:	

Proxy and Registration

Proxy:	10.99.31.140		
Outbound Proxy:			
Use Outbound Proxy:	no	Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	3600	Ans Call Without Reg:	no



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

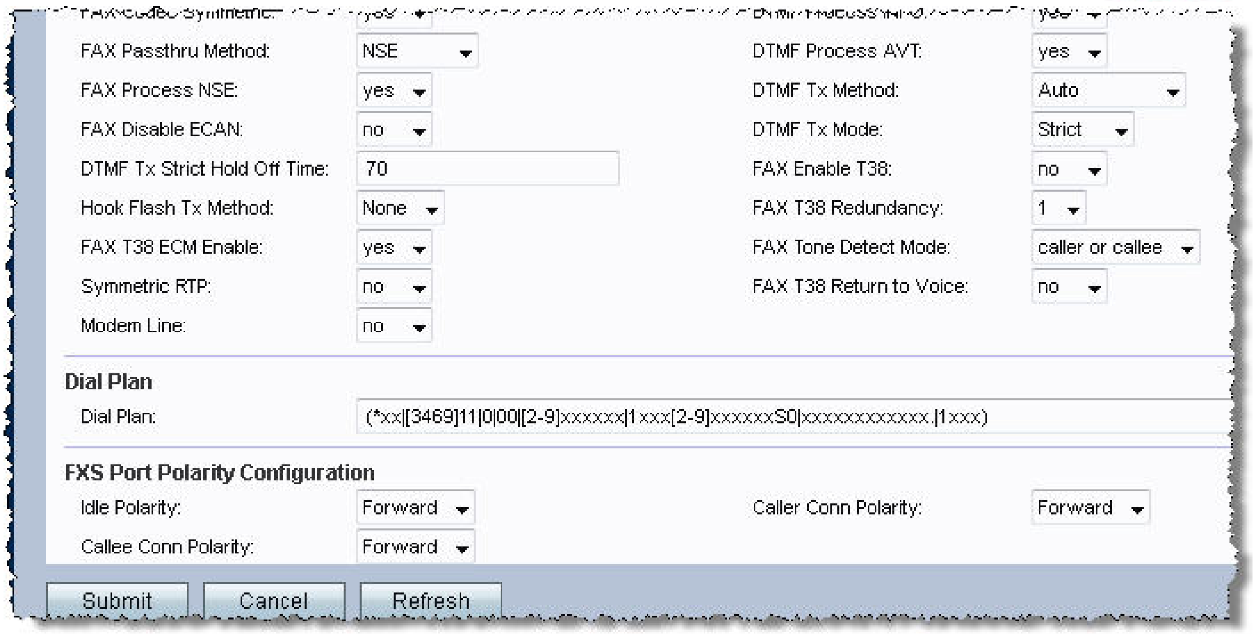
Display Name:		User ID:	1022
Password:	*************	Use Auth ID:	yes
Auth ID:	1022	Resident Online Number:	
SIP URI:			

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	MWI Serv:	yes
VMWI 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



c. Click Submit

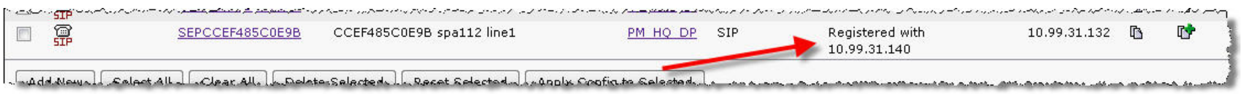
This completes the minimum SPA112 configuration required to register to a CUCM.

Verifying Registration Status

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

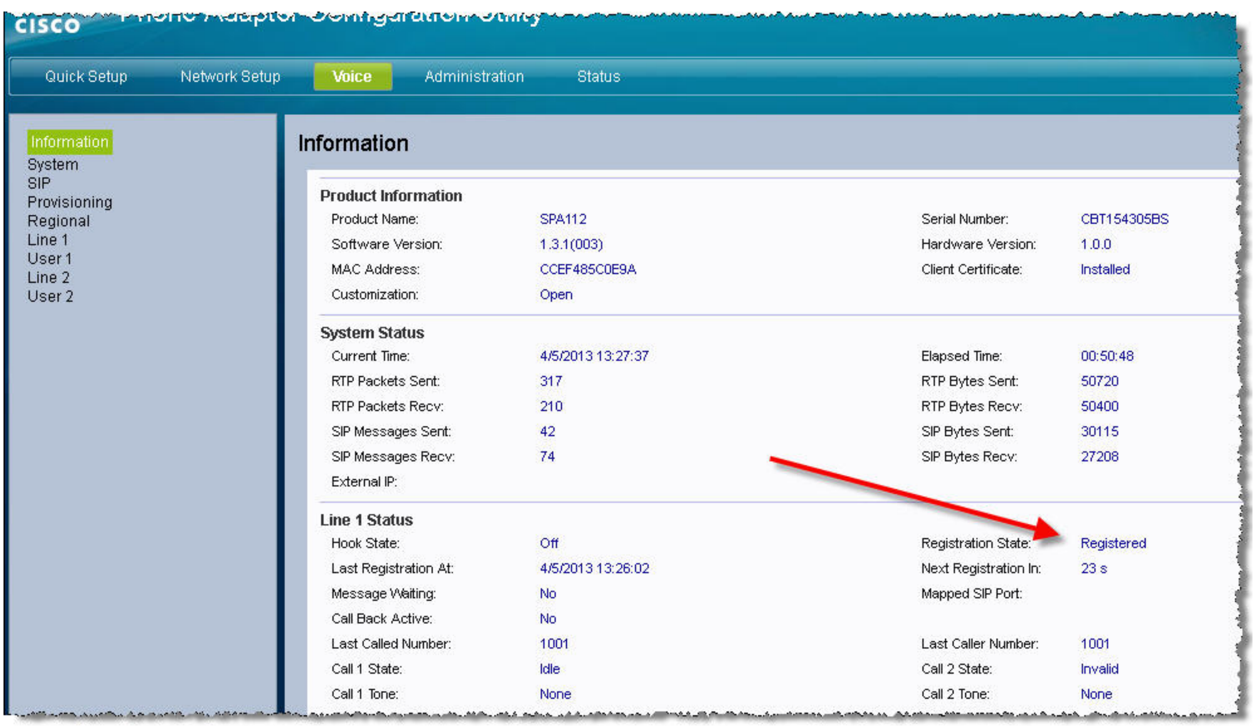
CUCM

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
Info
    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
Info
    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
Info
    328 69.457   14:36:55.519 10.99.31.140          10.99.31.132          SIP      472
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
Info
    329 69.458   14:36:55.520 10.99.31.132          10.99.31.140          SIP      647
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
Info
    332 69.459    14:36:55.521 10.99.31.140          10.99.31.132          SIP      349
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
Info
    333 69.460    14:36:55.522 10.99.31.140          10.99.31.132          SIP      472
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
Info
    336 69.472    14:36:55.534 10.99.31.132          10.99.31.140          SIP      827
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
Info
    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
Info
    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"

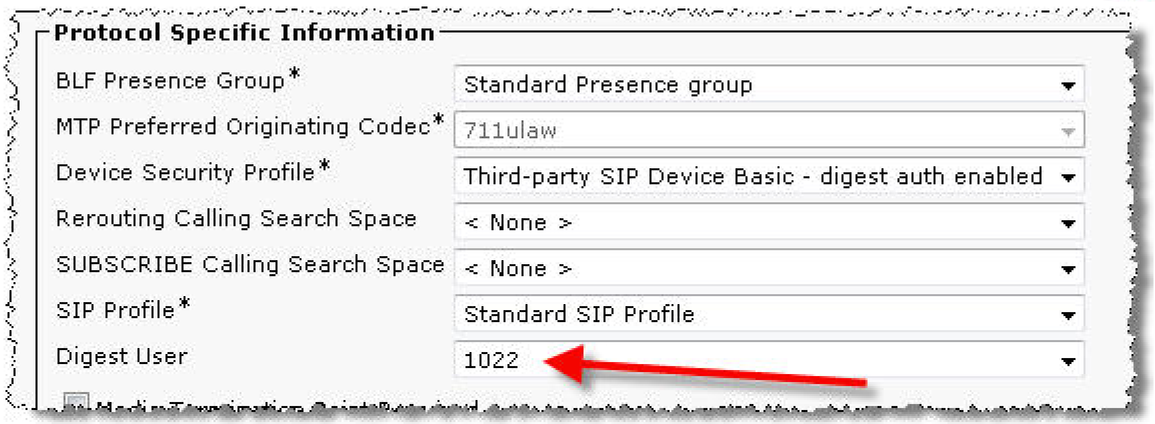
The SPA112 fails to register with the CUCM. Monitoring the network interaction and filtering for SIP, you see a SIP status message of 404 Not Found

Looking at the SIP Message Header, you see:

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

Possible Solution

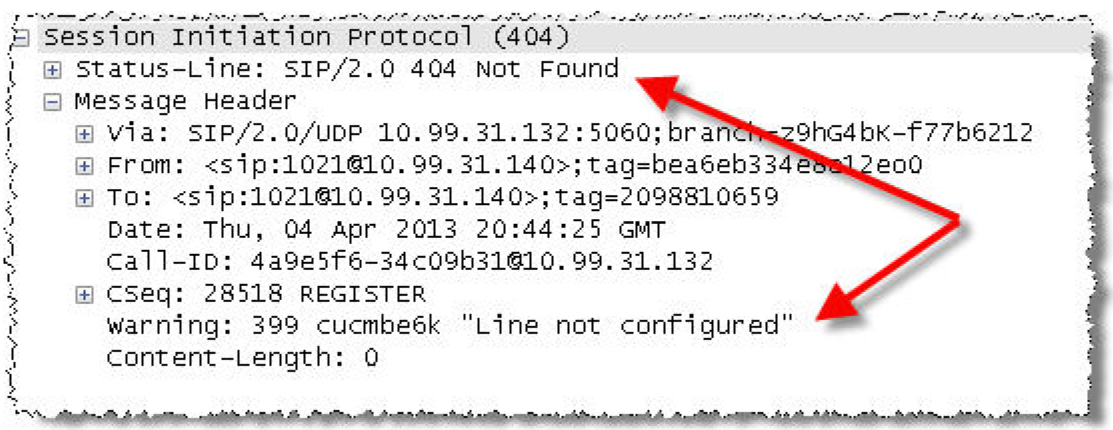
Define the 1022 user as the phone Digest User at Device > Phone > Find > Click the SPA112 Line 1 Link > Protocol Specific Information > Digest User: 1022



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:

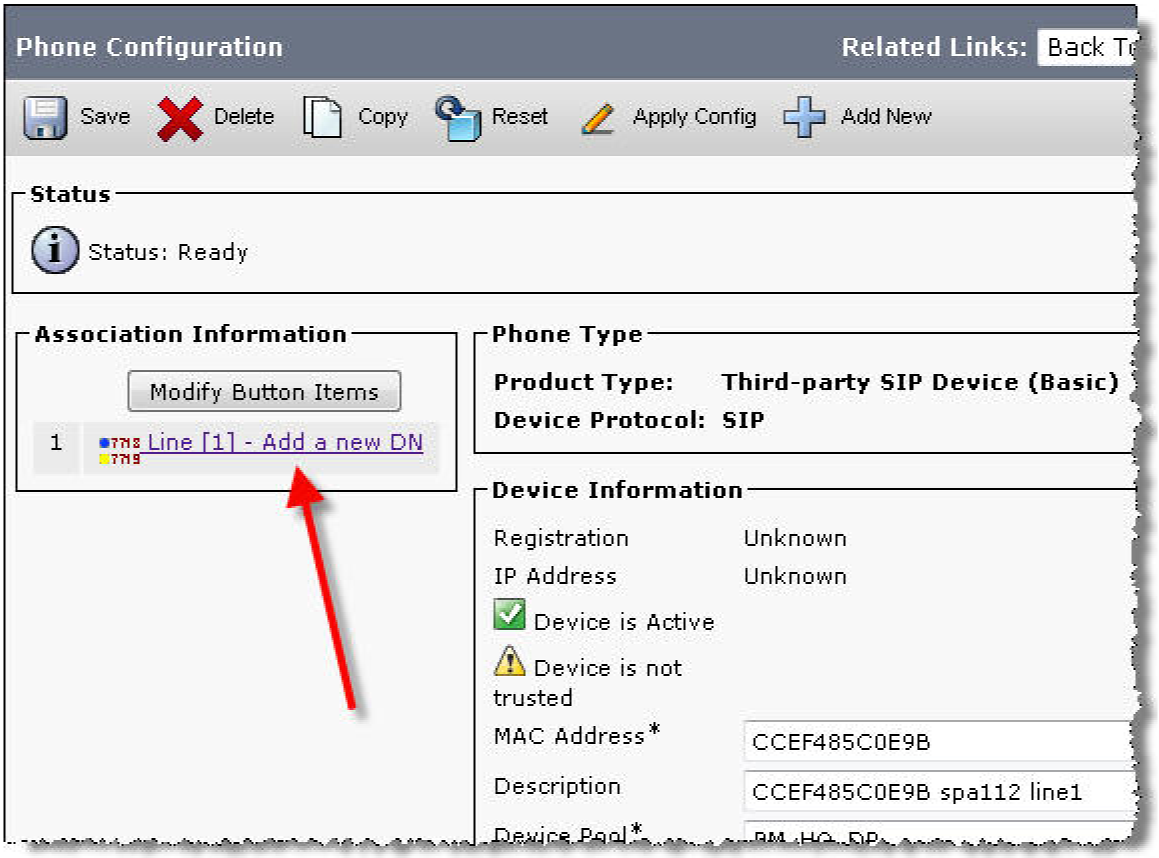
Looking at the SIP Message Header, you see:



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

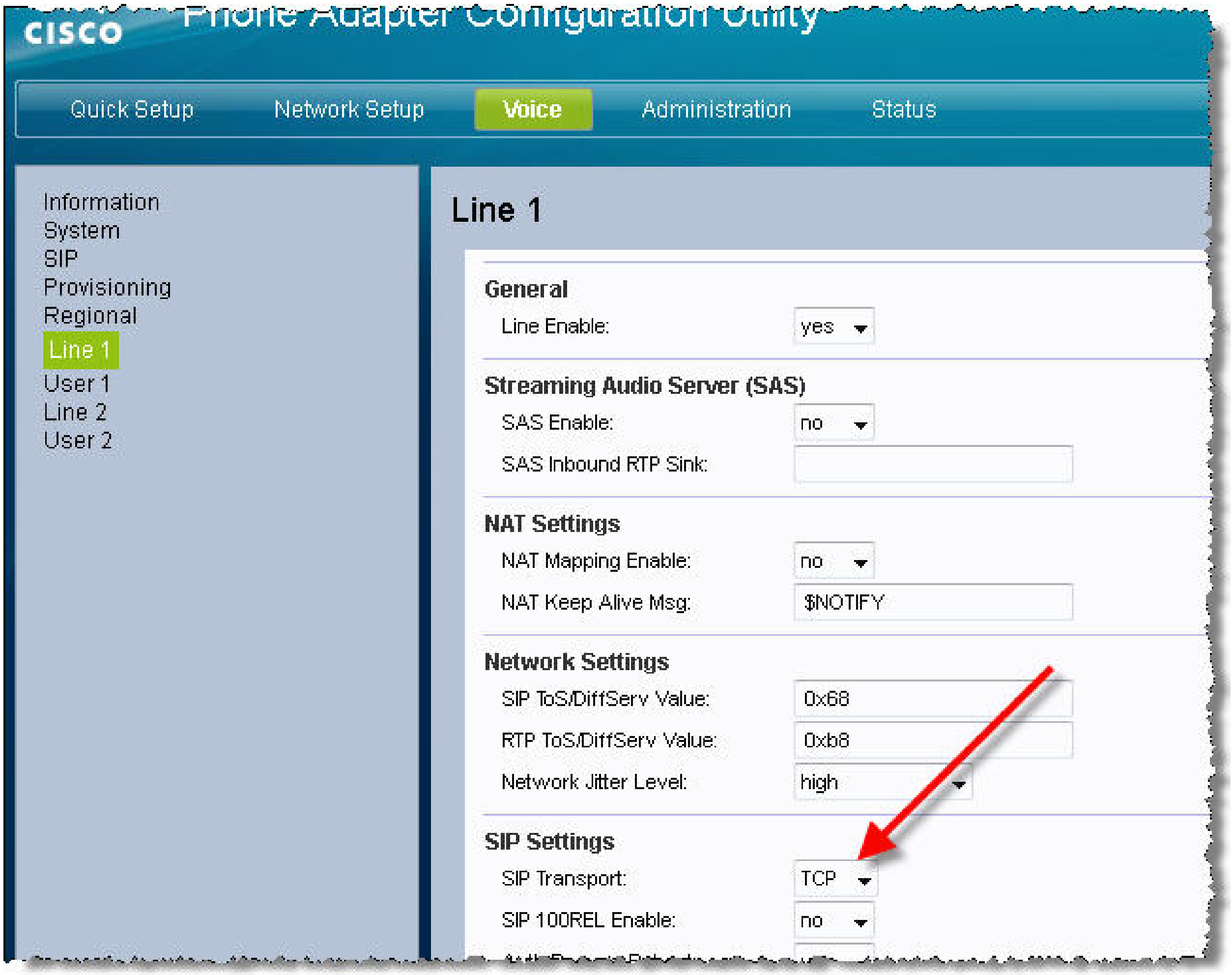


Outbound Call Fails with "Call cannot be completed as dialed" Message

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>