SPA232D& SPA302D—From Getting Started to Advanced Call Routing Scenarios

Contents

Scenario:

You've just unboxed your SPA232D ATA & DECT base station and optional SPA302D DECT handsets. This document's goal is to get your ATA and DECT solution up and running with as little effort as possible.

NOTE: This document includes links and is intended to be read electronically instead of being printed.

Overview:

Nothing in this document is unique to this document. All the information is available in some form spread across official Cisco documentation and the Cisco Community documentation and application notes. The goal of this document is to help you take the shortest path possible to using your new SPA232D ATA and DECT base station with your SPA302D handsets.

The SPA232D is a DECT base station, an analog telephone adaptor (ATA), and a gateway device:

1. The SPA232D DECT base station side allows you to optionally register up to 5 Cisco SPA302D handsets. [You do not have to connect any if you have no use for DECT handsets]
2. The SPA232D also allows you to optionally use an analog phone or fax machine connected to its PHONE (FXS) port. [You do not have to use this port if you have no need]
3. The ATA side allows you to use an old style analog phone to make or receive phone calls over the Internet. [You do not have to use this feature if you have no need]
4. The gateway side allows you to make or receive phone calls from the public switched telephone network (PSTN) connected to your house wiring via the LINE (FXO) port. [You do not have to use this port if you have no need]
5. You can also combine any or all of the above features at the same time if you have a need.

Getting Started

1. Connect the SPA232D 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 SPA232D Quick Start Guide located here.
2. Access the web-User Interface (web-UI) at the **http://spa232dIPaddress** URL.
   The default administration credentials are admin / admin

Registering SPA302D DECT Handsets

Be sure to register your SPA302D handsets to the SPA232D base station as soon as possible. An unregistered handset uses a lot of power attempting to locate a DECT base station.

1. Register by long-pressing (longer than 7 seconds) the button on the SPA232D base station. The LED in the base station's button must flash rapidly to indicate registration mode. A shorter than 7 second button-push results in the SPA302D entering paging mode, press it again to exit paging mode and then long-press to reach registration mode.

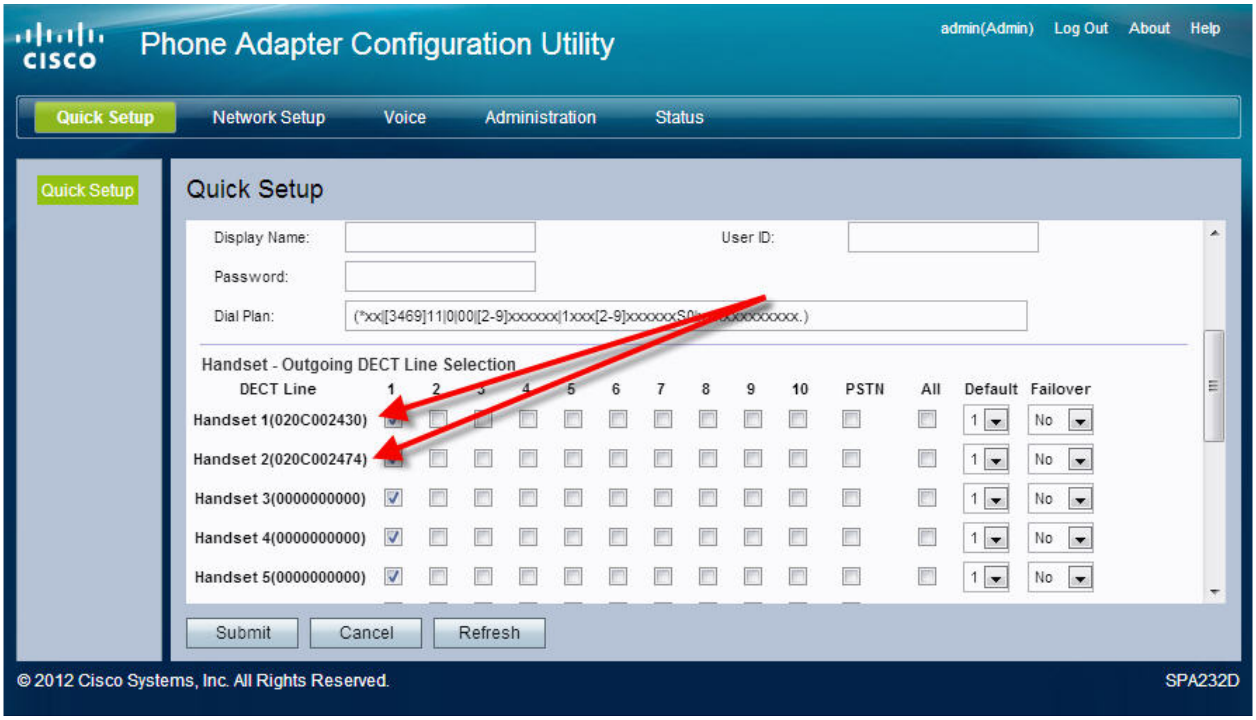
2. On SPA302D: **Menu > Settings > Handset Registration > Register > Confirm** [leave PIN blank]. The handset will register and display a Handset number near the top-right of the handset's display.

Understanding Handset *N* (*N*)

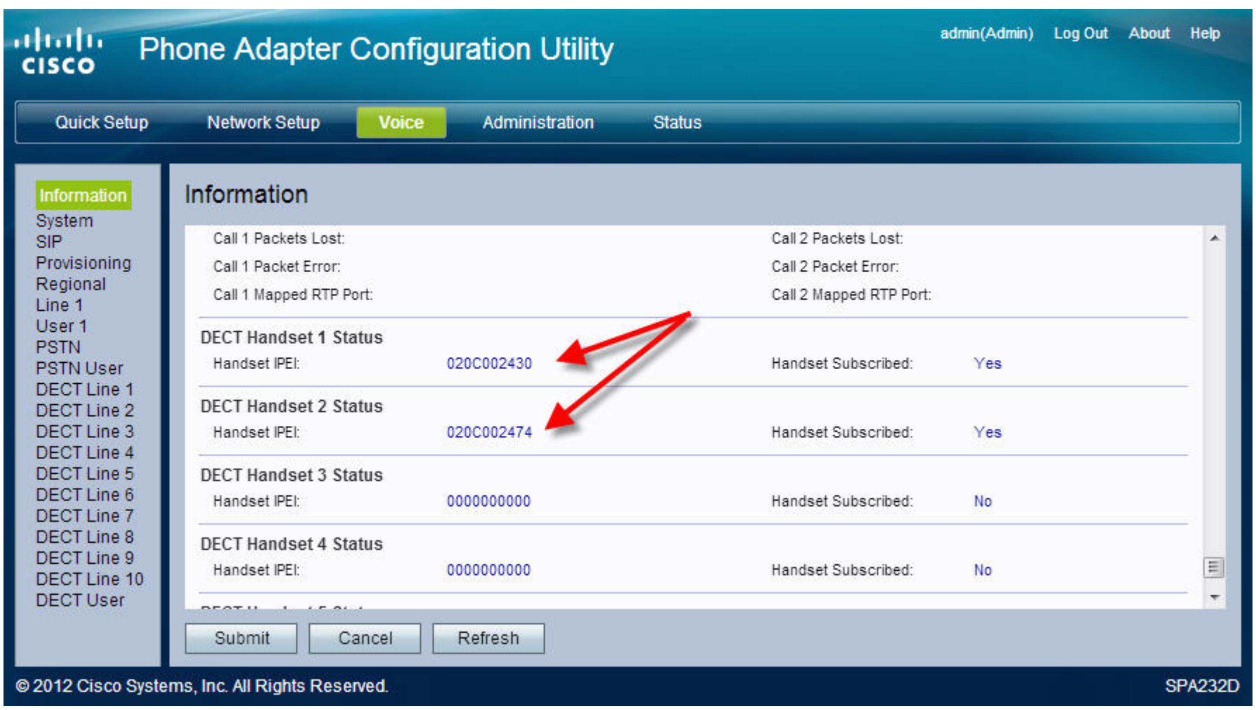
The Handest 1 (1) near the top-right of the SPA302D's display identifies this handset's identity with the SPA232D DECT base station.

This is useful if you want to configure specific features for this handset. You can locate this handset in the SPA232D by its Handset number or IPEI# (International Portable Equipment ID) at the:

- Quick Setup tab:



- Voice tab > Information > DECT Handset N Status:



Updating the SPA232D Firmware

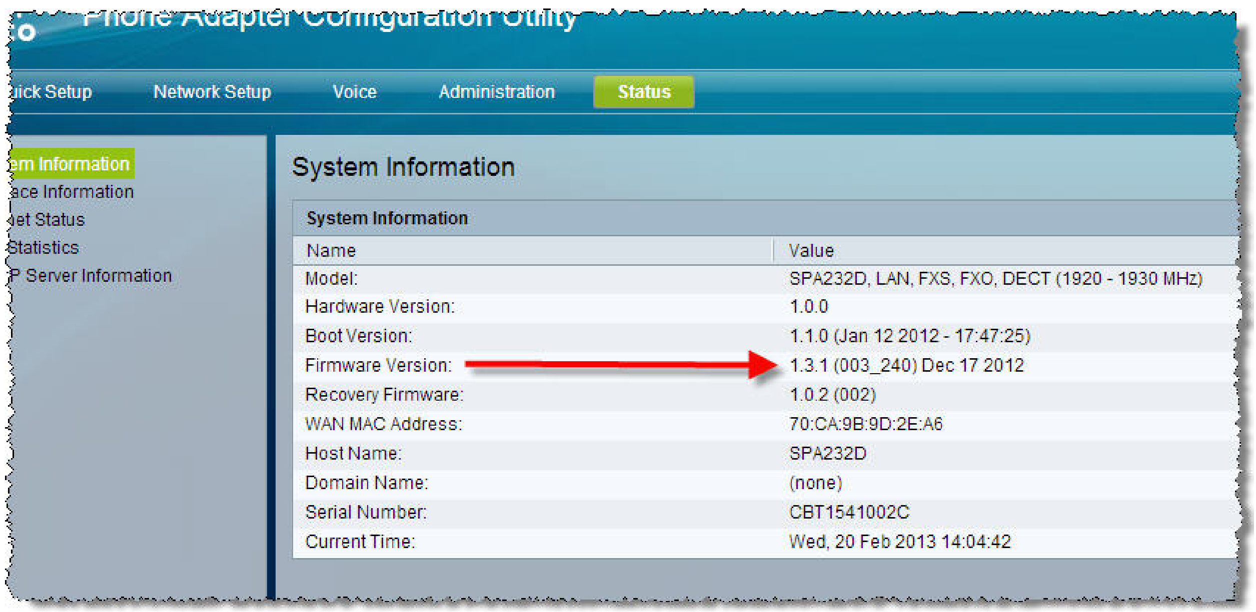
Update the firmware on your SPA232D to the most current firmware available from [here at Cisco] because newer firmware may have been released since your ATA was manufactured.

Determining the SPA232D Installed Firmware Version

The https://supportforums.cisco.com/docs/DOC-29465document describes how to determine the firmware version running on your ATA.

The SPA232D uses different firmware to other SPA1x2 devices because it also supplies firmware to DECT handsets that register to it. This is why you will see a firmware version similar to this:

The numbers in the parenthesis describe first the SPA232D and then the SPA302D sub-versions. For example the SPA232D is running 1.1.1 subversion 003 for itself and subversion 240 for any registered SPA302D handsets.



Locating Current SPA232D Firmware

Navigate to the **http://cisco.com/ > Downloads Home > Products > Voice and Unified Communications > Communications Infrastructure > Voice Gateways > Cisco Small Business Voice Gateways and ATAs > Cisco SPA232D Multi-Line DECT ATA** page.

Alternatively, a shortcut to the Download Software page for the SPA232D is located here

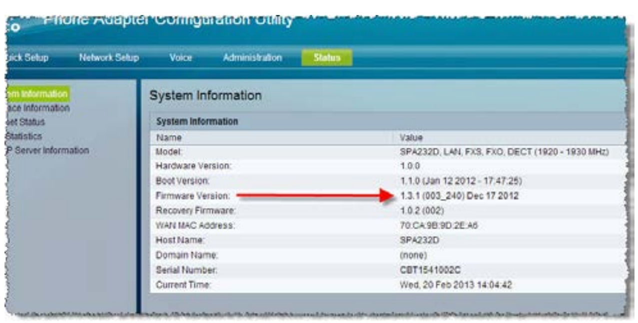
How do I Update the Firmware on my SPA232D ATA?

The https://supportforums.cisco.com/docs/DOC-29477 document describes how to upgrade the firmware on your ATA.

What Version of Firmware is on my DECT Phones:

You can view the version of firmware installed on the SPA302D by pressing the center (menu) button and then using the arrow keys to navigate to the Settings option.
[**Menu > Settings > Phone Info > Software Version**]

Notice that the software version 2.40 is the same as the last number in the software version reported by the SPA232D from the earlier section. This means that the firmware on the handset is current. The only way to upgrade the handset is to first upgrade the firmware on the SPA232D base station.



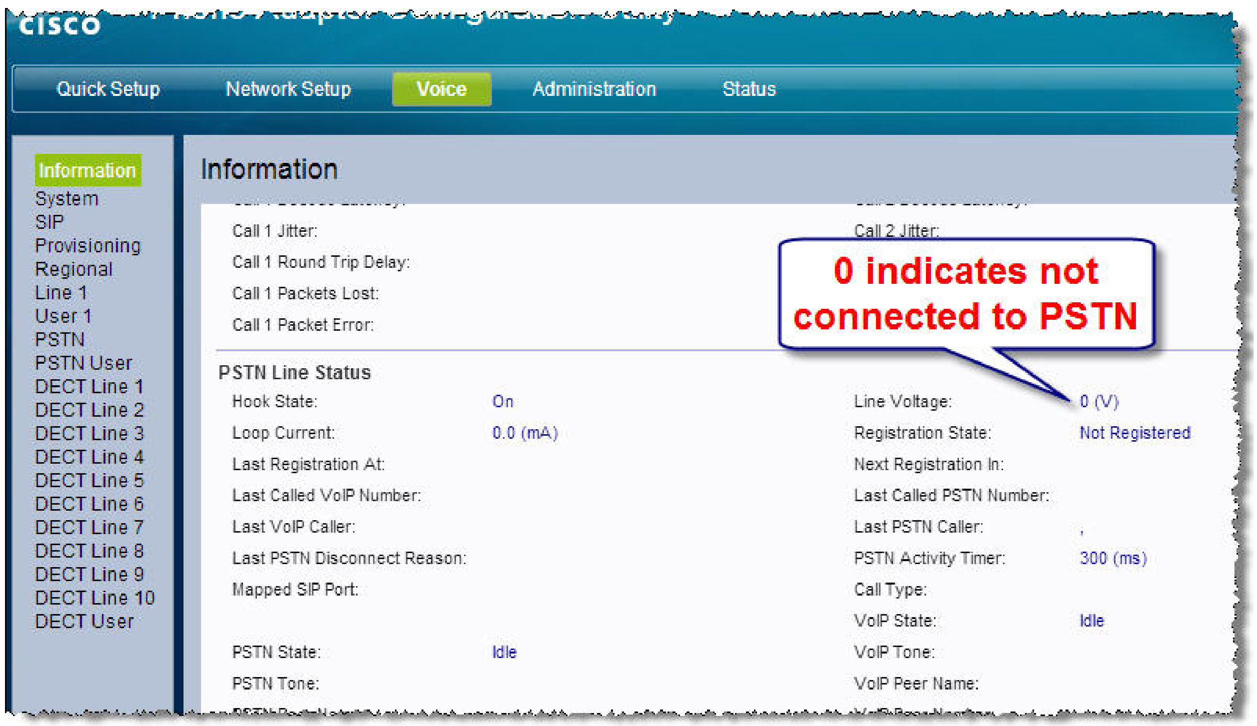
How do I Update the Firmware on my SPA302D Phone?

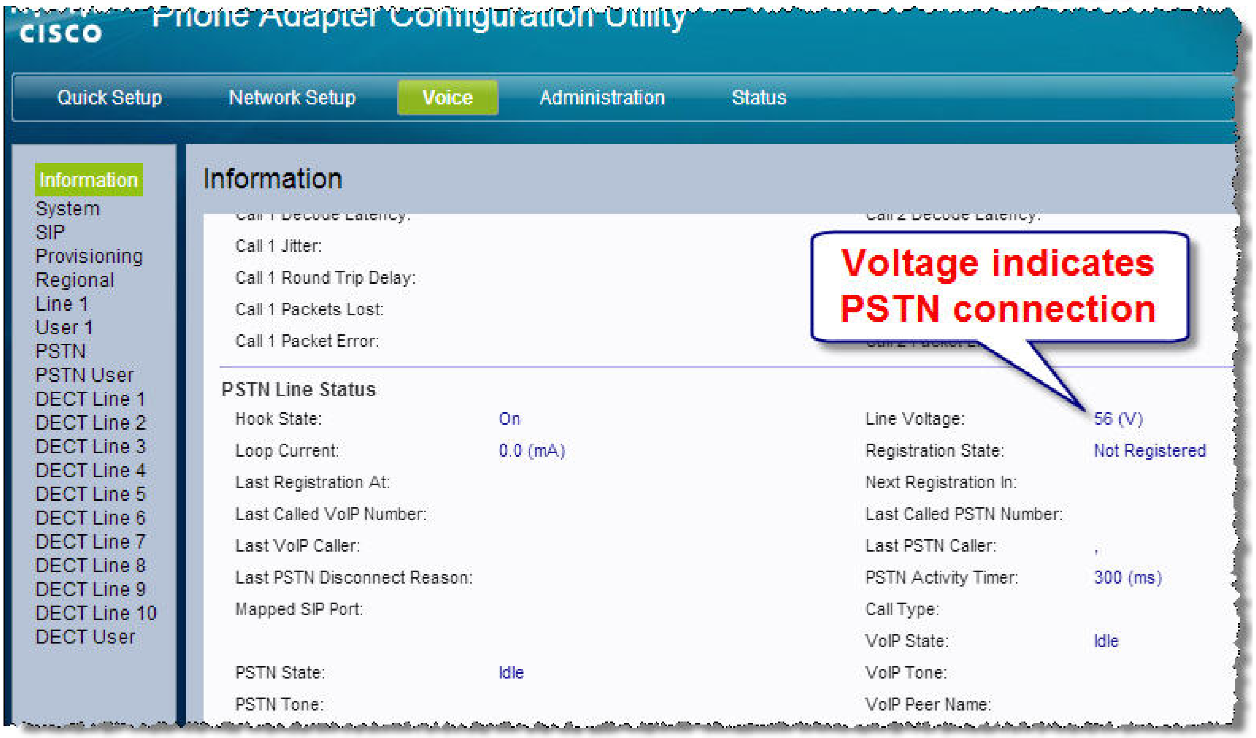
The firmware version of your SPA302D DECT phone is controlled by the SPA232D DECT base station to which the SPA302D is registered. You can initiate a firmware upgrade as follows:

Press **Menu>Settings > Update Software > Confirm Check for software update > Select** if you want to upgrade to the displayed version, else press Cancel.

Configuring the LINE (PSTN) Port

You can connect a PSTN [public switched telephone network] line, for example, the RJ-11 wall jack in your house's wiring, to the SPA232D. This will allow phone calls to be made out via this PSTN line.
Before making any connections, make sure that the house's phone jack is operational by first connecting a regular analog (old style phone) and verifying that you can make and receive calls.
Connect the SPA232D's LINE port to the house's PSTN wiring.





LINE (PSTN) Calls

There are two categories for PSTN calls, inbound and outbound.

Inbound PSTN Calls

Inbound PSTN calls can be routed to any or all of the following:

- Analog phone connected to the PHONE port
- SPA302D DECT Handset 1 Through 5

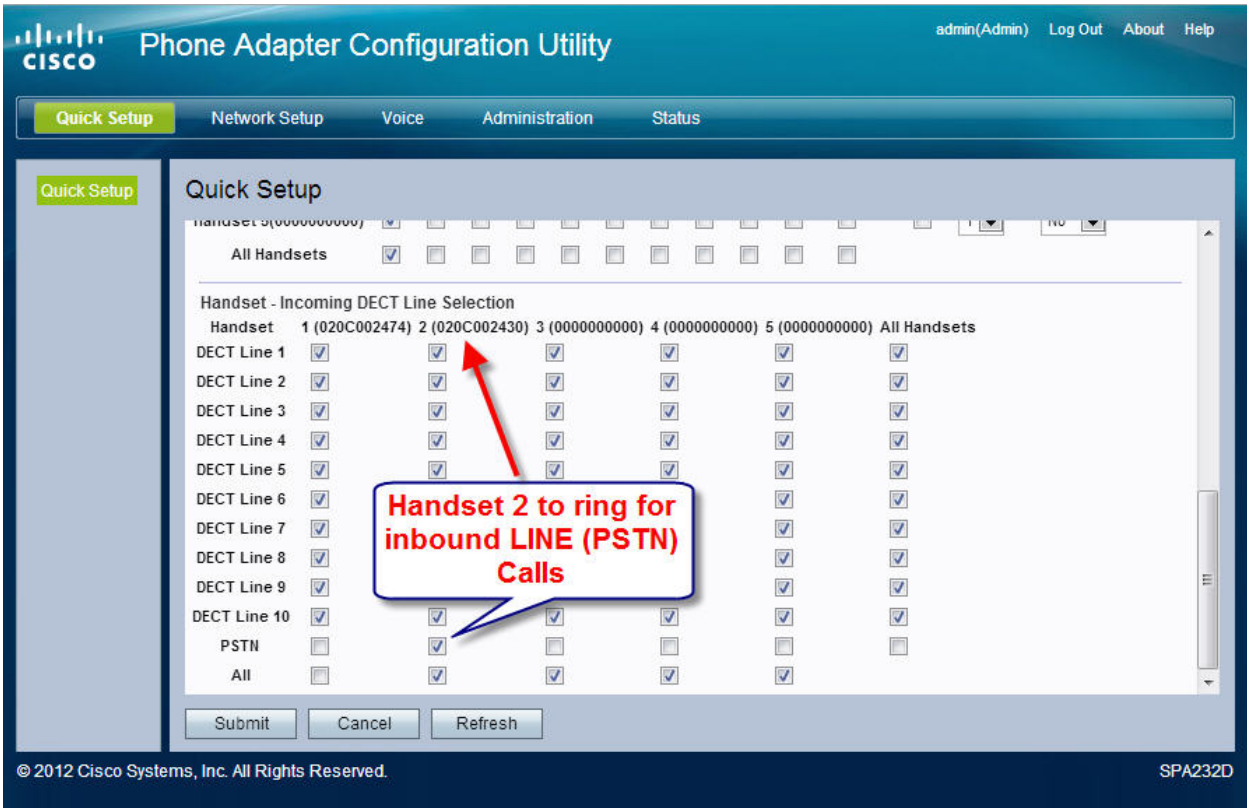
Configuring the PHONE Port to Receive Inbound PSTN Calls

1. Connect a working PSTN line to the SPA232D LINE port
2. Connect a working analog phone to the PHONE port of the SPA232D

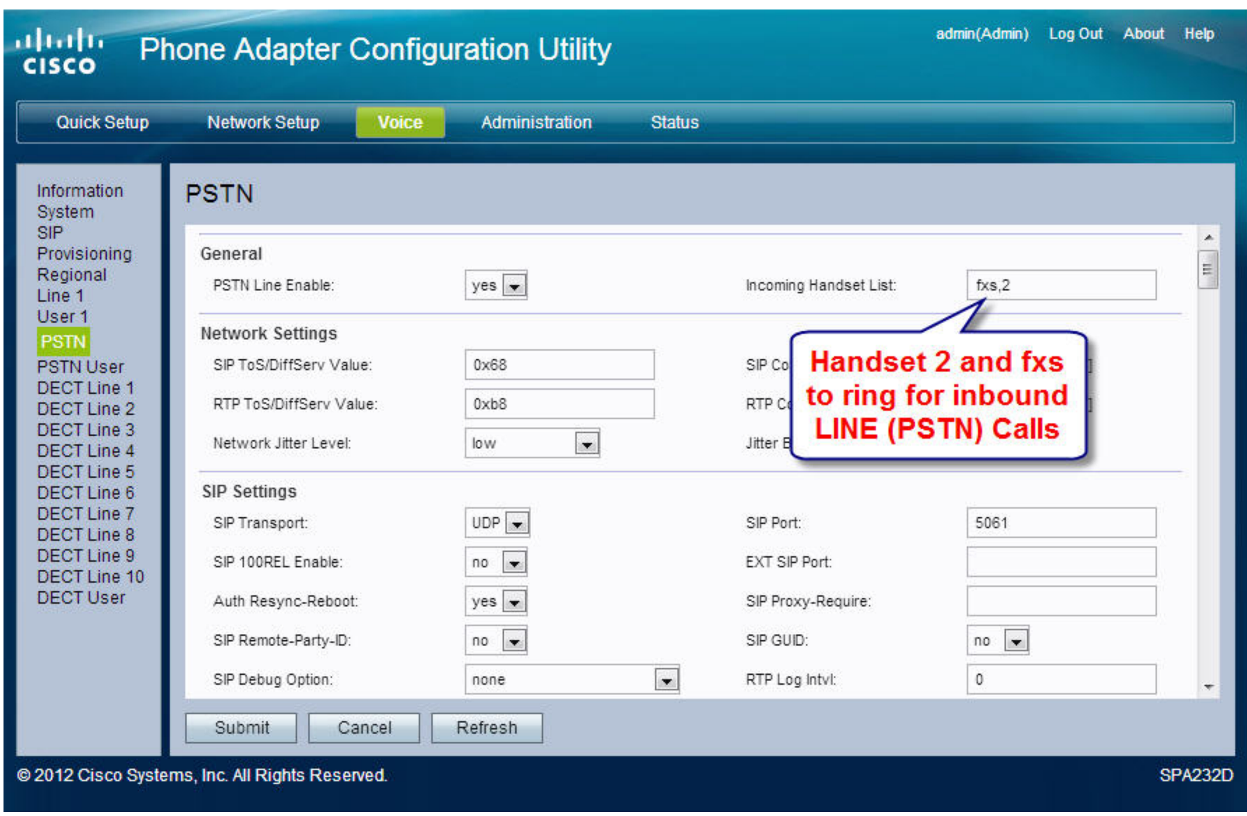
When an outside party dials the PSTN number, the analog phone connected to the PHONE port will ring.

Configuring the SPA302D DECT Handsets 1-5 to Receive Inbound PSTN Calls

1. Connect a working PSTN line to the SPA232D LINE port
2. Register the SPA302D handset to the SPA232D base station. Once registered, the SPA302D handsets will automatically ring for any inbound PSTN call received on the SPA232D's LINE port.
3. You can configure which SPA302D handsets must ring for inbound PSTN calls. This example will configure Handset 2 to ring in addition to the analog phone connected to the PHONE port of the SPA232D using either the Quick Setup tab or the Voice tab:
   - Using the **Quick Setup tab > Handset - Incoming DECT Line Selection** >select PSTN under Handset 2 as shown below



- **OR**: Using **Voice tab > PSTN > General > Incoming Handset List**: fxs,2



Outbound PSTN Calls

Outbound calls to the PSTN (LINE port of the SPA302D) can originate from any or all of the following: analog phone connected to the PHONE port of the SPA302D, or registered Handsets 1 through 5.

Configuring the PSTN Port to Allow Outbound Calls from Analog Phone Connected to the PHONE Port

If you have not configured anything and go off-hook on the analog phone connected to the PHONE port, you will hear dial tone and will be able to make outbound phone calls just like you would if the analog phone was connected directly to the wall jack in your house.

NOTE: Once you configure a SIP Proxy for the PHONE port, all calls from the analog phone connected to the PHONE port will be routed out via the specified SIP Proxy. You can force the use of the LINE (PSTN) port by using the **<:@gw0** routing string in the Phone 1 dial plan as described later in the Making an Outbound Call from Analog Phone Using LINE (PSTN) section

NOTE: If your SPA232D is disconnected from power, you can still make and receive calls with the analog phone connected to the powered down SPA232D.

Configuring SPA302D DECT Handset 1 to Make Outbound Calls from the PSTN Line

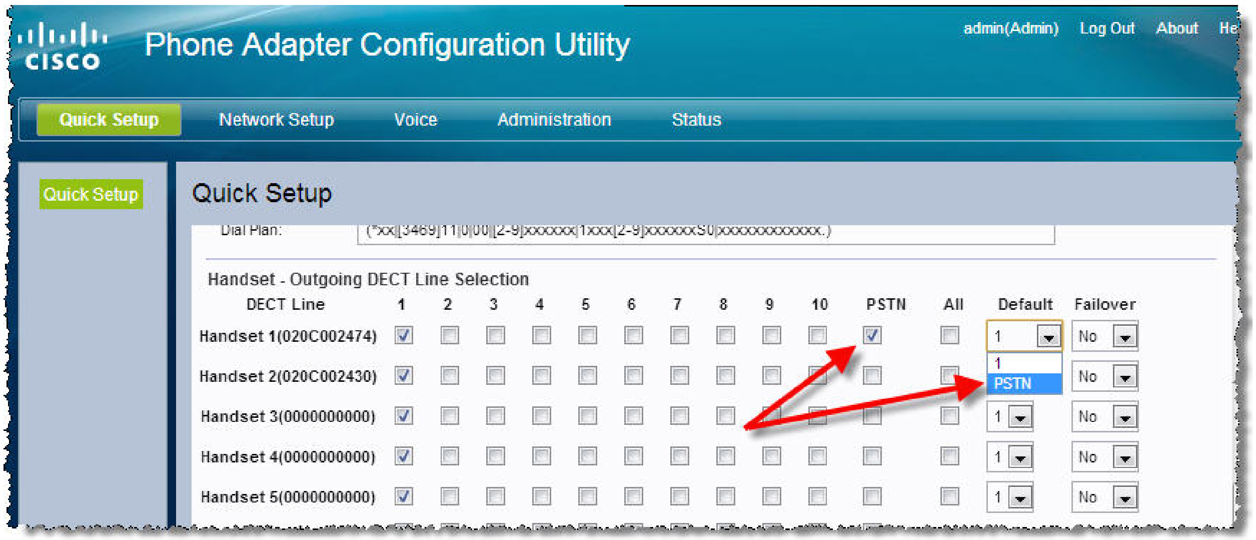
If you have not configured anything for the SPA302D DECT handset, you cannot use it to make calls via the PSTN line or via any SIP proxies.

NOTE: The SPA232D is not a PBX so you cannot make calls between phones connected to it unless you make an outbound call to the target phone using an external line.

By default, the DECT Line 1 line is assigned as the SPA302D's outbound default line regardless of whether DECT Line 1 is available, registered, or configured. You can change the SPA302D's outgoing default line to the PSTN using one of the following two methods:

A. Using Quick Setup

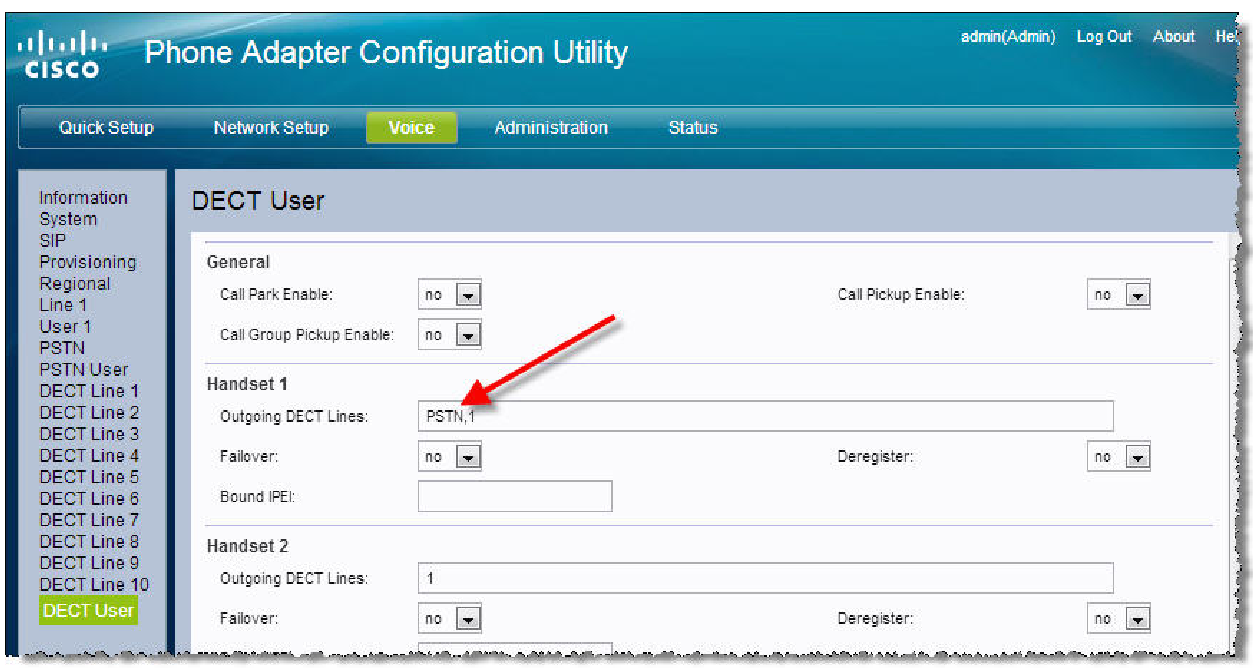
1. SPA232D web-UI > Quick Setup tab
2. Click PSTN to make it available as an outgoing target in the Default dropdown menu
3. Change Handset - Outgoing DECT Line Selection from Default of 1:



4. Click Submit

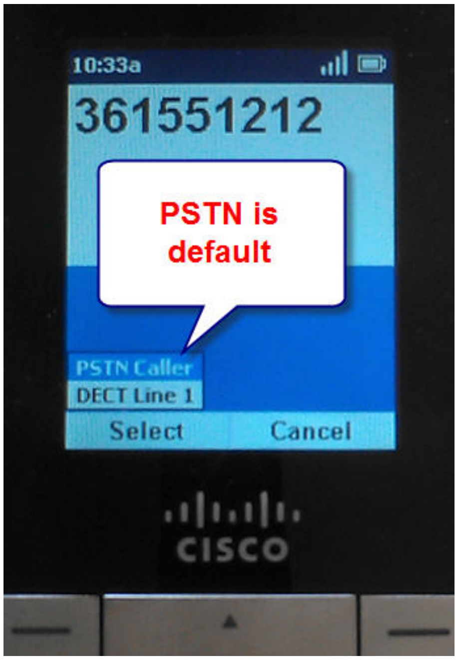
B. Using **Voice tab > DECT User > Handset 1 > Outgoing DECT Lines**:

1. For Handset 1, change Outgoing DECT Lines by inserting PSTN as follows:



2. Click Submit

Once you've completed one of the above tasks, you are presented with a popup offering available outgoing lines with the default line already selected when dialing:



Viewing Status of Default Outgoing Line

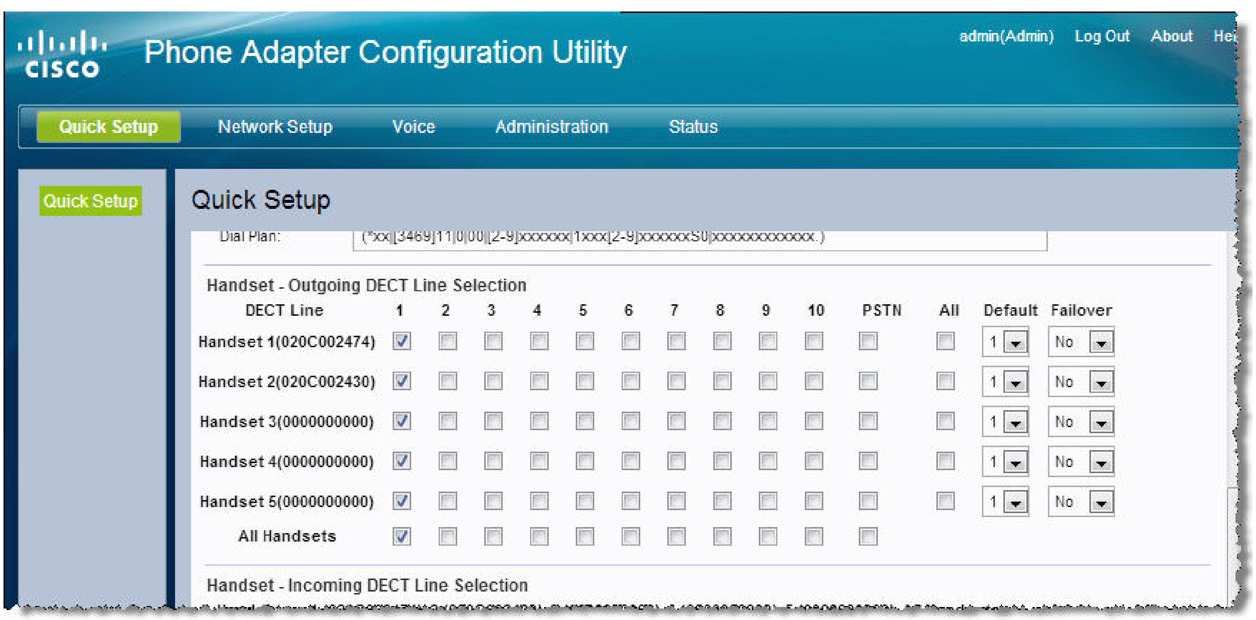
You can view the default outgoing line's status as follows: **Menu button > Call Settings > Line Status**
A status of Not Ready indicates that you cannot make or receive calls from this line.

SPA302D Handset's Default Outgoing Line

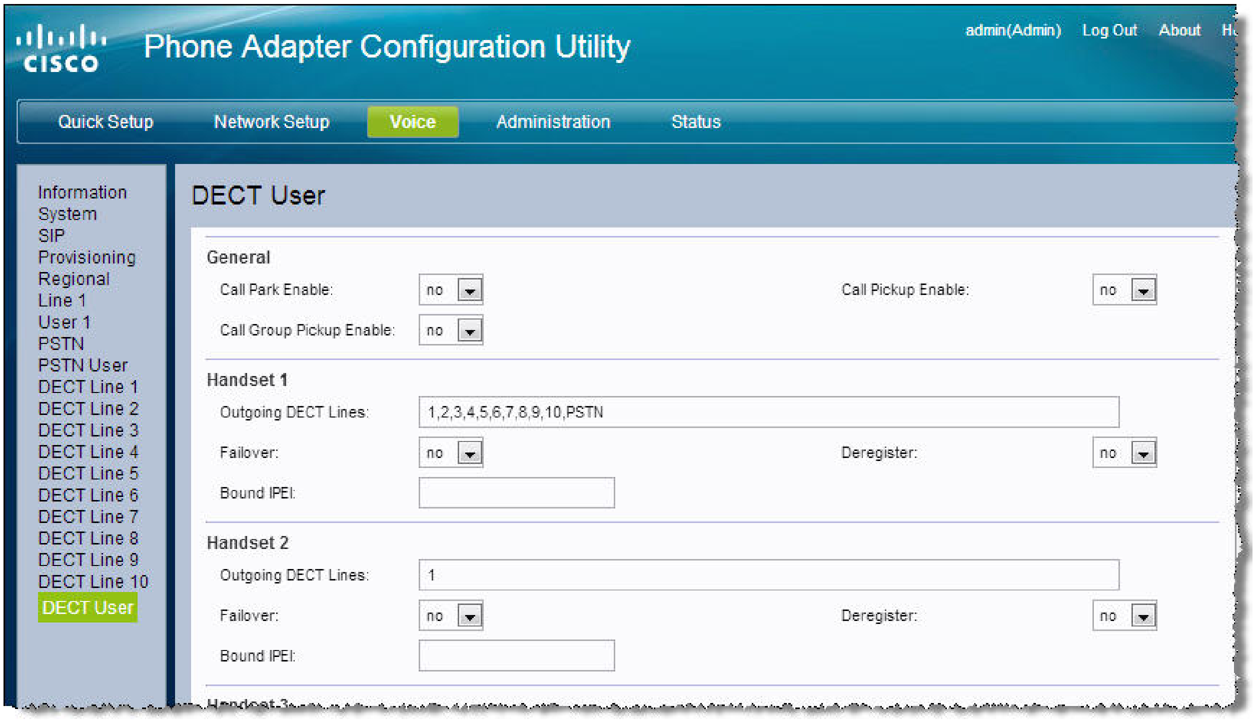
View the SPA302D handset's default outgoing line as follows:
Menu button > Call Settings > Outgoing Default Line

You can also view and configure the outgoing lines using the SPA232D's web User Interface (web-UI):

Quick Setup tab:



Voice tab > DECT User:



Using VoIP

This section relates to configuring the SPA232D so that you can use an Internet Telephony Service Provider (ITSP) to make and receive calls over the Internet using either an old-style (analog) phone or fax machine connected to the PHONE (FXO) port of the SPA232D or using a registered SPA302D DECT handset. Similar to PSTN calls, there are two categories of calls to consider, inbound and outbound.

This section assumes that you have already signed up with an ITSP and have received the following information which you will use:

- SIP Proxy (proxy)
- SIP Outbound Proxy (optional)
- User ID (sometimes the phone number is used)
- Password
- Authentication ID (optional auth ID)

Without the above from an ITSP, you cannot configure voice over IP (VoIP).

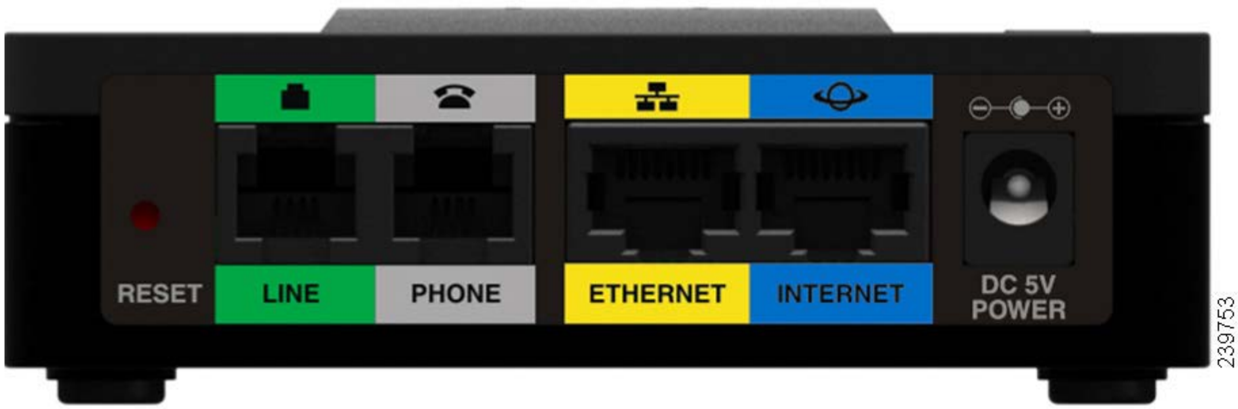
NOTE: Your SPA232D can handle multiple ITSP accounts if you desire. Each of the following can have their own ITSP account allowing for up to 12 different ITSPs or multiple accounts from different ITSPs:

1. Line 1
2. PSTN

3. DECT 1-10 [10 accounts]

NOTE: The SPA232D is capable of sharing accounts but some ITSPs block this capability.

The SPA232D has several ports at the back.



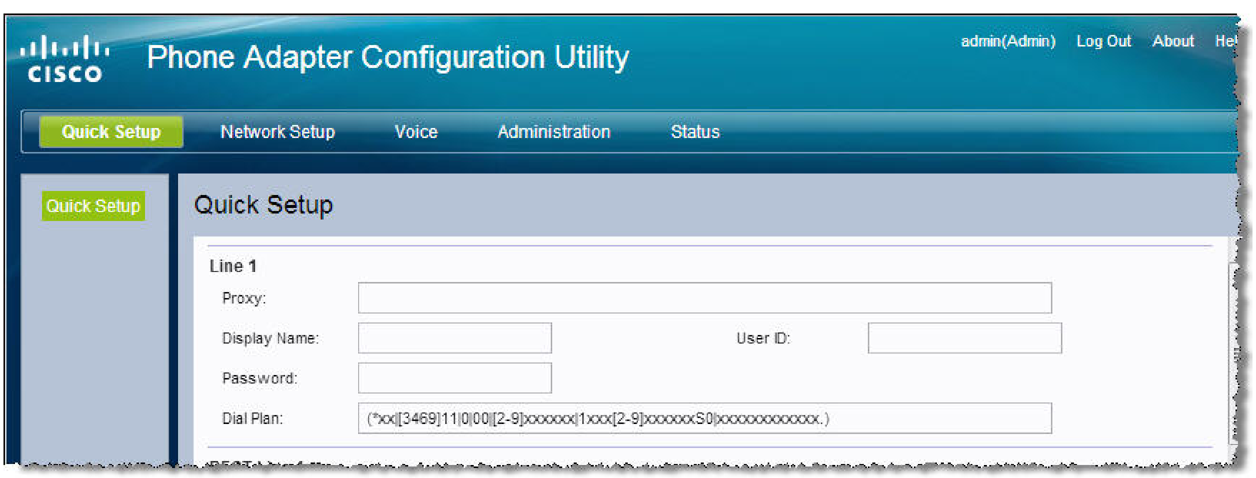
The naming can be a little confusing so here's a different view:

- LINE (FXO) <> PSTN in web-UI<> PSTN line to house wiring [**<:@gw0>** route string in dial plan]
- PHONE (FXS) <> Line 1 in web-UI<> analog phone or fax machine

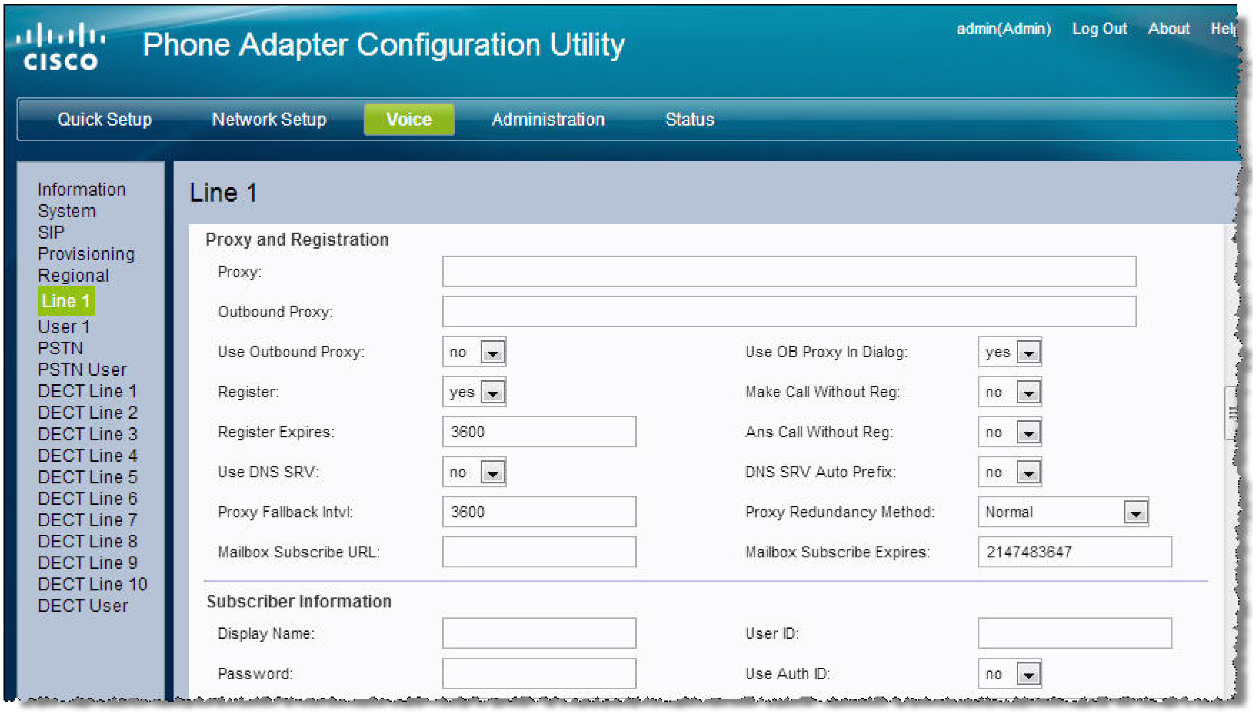
Configuring Line 1 (FXS port SIP Registration)

Follow this section to configure the PHONE (FXS) port of the SPA232D so that devices connected to the PHONE port can make and receive calls using an Internet Telephony Service Provider (ITSP).

You can configure Line 1 using either the **Quick Setup tab > Line 1** or the **Voice tab > Line 1**. The two pages are interlinked, once you submit the information, the other page's data fields are populated. The Quick Setup page has all the relevant fields close together for faster population.

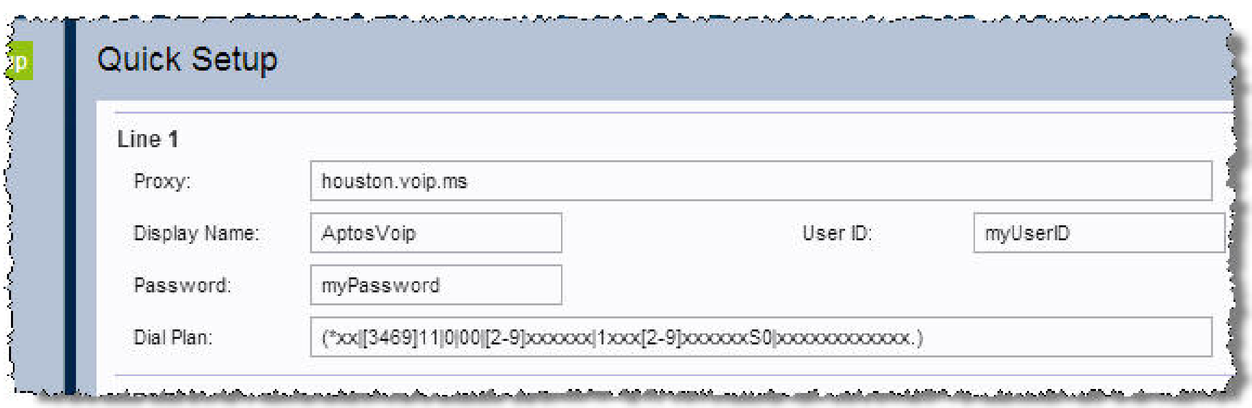


Or, use the parameters in the **Voice tab > Line 1 > Proxy and Registration:**
and then the **Voice tab > Line 1 > Subscriber Information:**

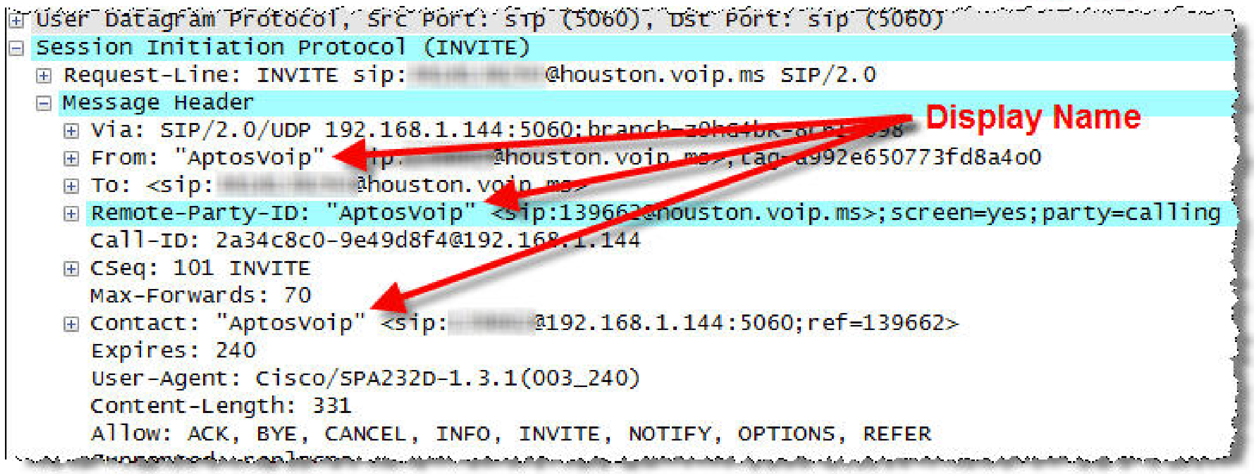


1. Enter the Proxy
2. Enter the optional Outbound Proxy
3. Enter the Display Name [optional, this shows up in the SIP Message Header]
4. Enter the User ID
5. Enter the Password
6. Click Submit

Once completed, the fields could look like this:

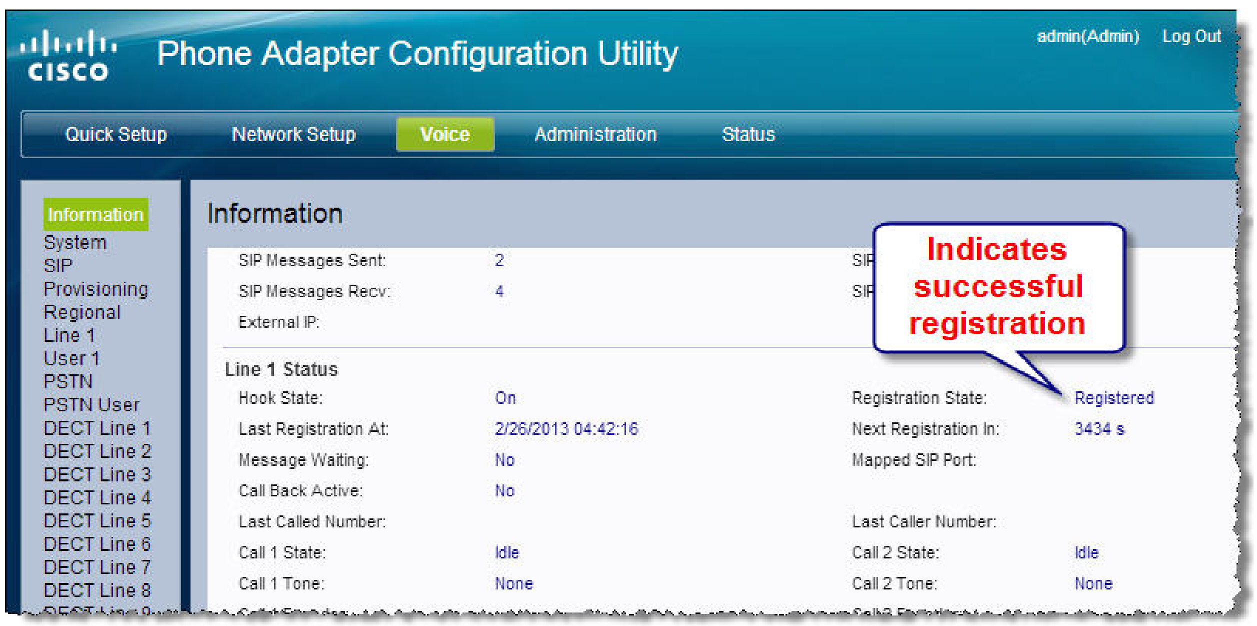


The optional Display Name information is displayed in the SIP Message Header and can be viewed with network packet capture tool such as Wireshark:



Verifying the Registration Status of Line 1

Once you've submitted your ITSP credentials, the SPA232D will restart and attempt to register with the specified ITSP. Verify the registration status with the web-UI at **Voice tab > Information > Line 1 Status:**



Inbound VoIP Calls

NOTE: Your ITSP controls the destination of the SIP INVITE based on their system and your configuration of your account/s. For example the voip.ms ITSP that I use in this document's configurations allows sub accounts where you can define which account or sub account must receive any incoming calls. Keep this in mind when trying to determine why certain targets may not ring for inbound calls.

Configuring the PHONE (FXS) Port to Receive Inbound VoIP Calls

Once Line 1 is registered to your ITSP, any inbound calls to the ITSP will make the analog phone or fax connected to the PHONE (FXS) port ring by default. If you registered Line 1 to your ITSP more than 10 minutes ago and the device has been idle you may receive a busy tone if the SPA232D is deployed in a network using NAT. Refer to the troubleshooting section describing NAT Keep Alive messages.

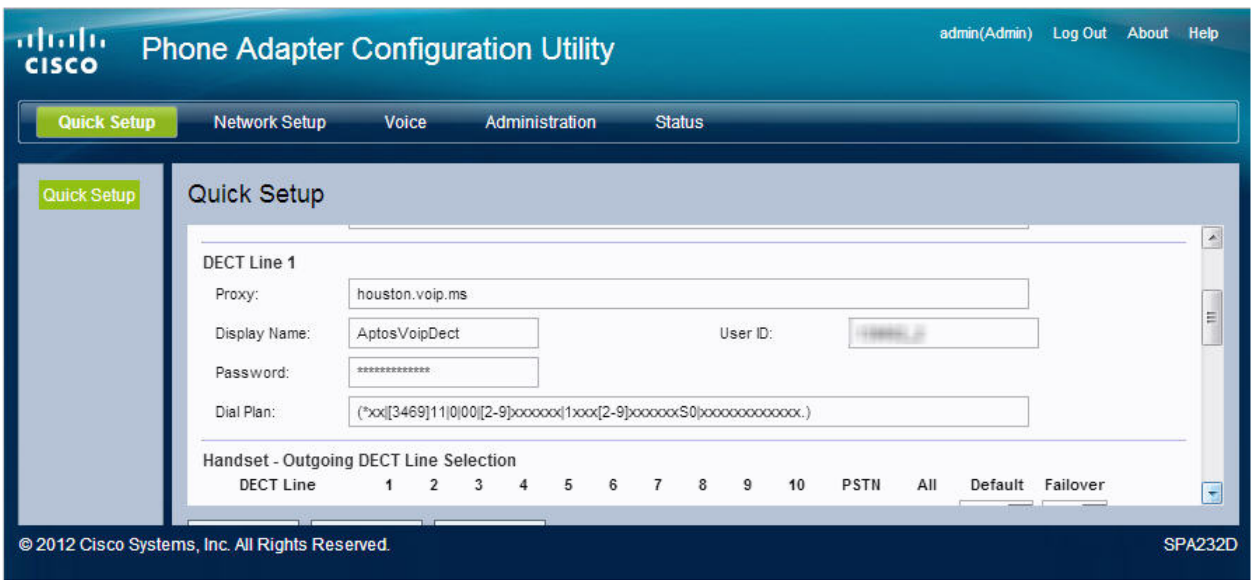
Configuring the SPA302D DECT Handsets 1-5 to Receive Inbound VoIP Calls

You must register each SPA302D Handset to an ITSP in order to make and receive calls.

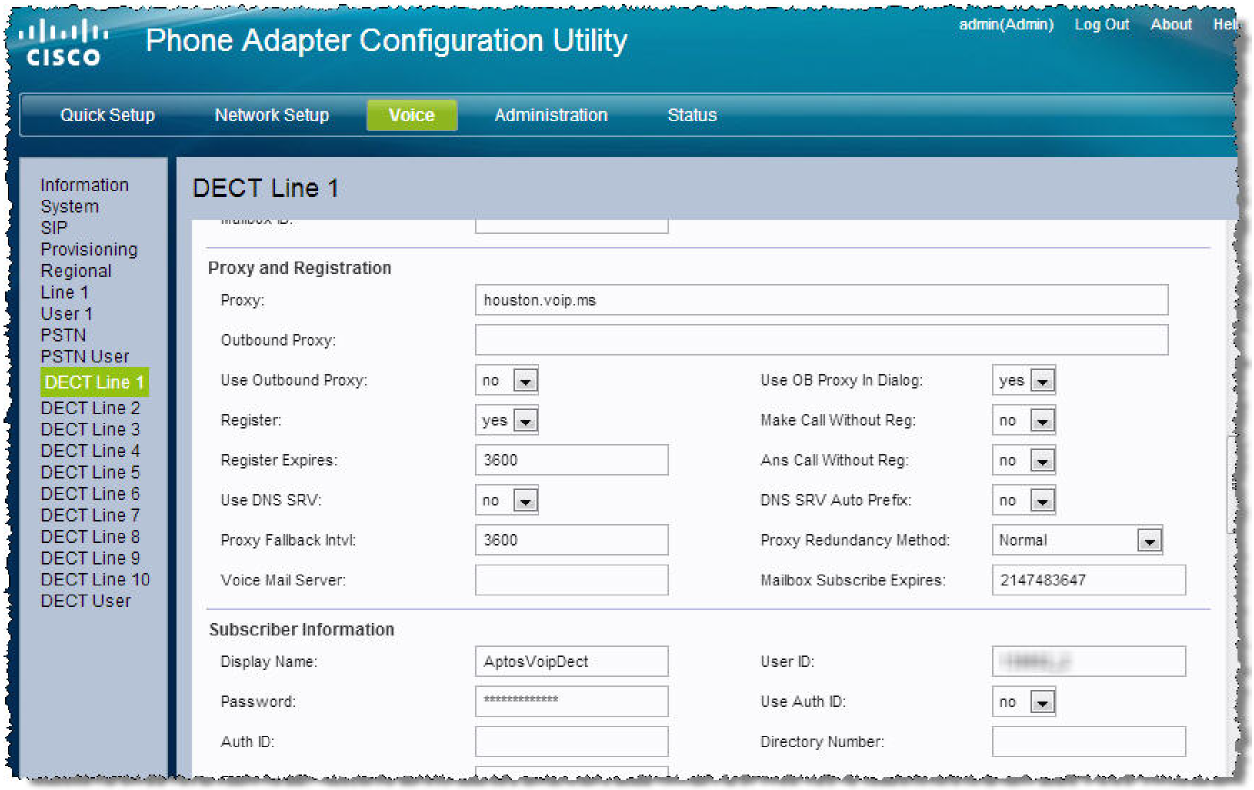
You can configure DECT Line N using either the **Quick Setup tab >DECT Line N** or the **Voice tab >DECT Line N**. If configuring DECT Line 1, the Quick Setup tab and the Voice > DECT Line 1 tab pages are interlinked so once you submit the information, the other page's data fields are populated. The Quick Setup page has all the relevant fields close together for faster population.

1. Enter the Proxy
2. Enter the optional Outbound Proxy
3. Enter the Display Name [optional, this shows up in the SIP Message Header]
4. Enter the User ID
5. Enter the Password
6. Click Submit

Once completed, the fields could look like this:



Or, use the parameters in the **Voice tab >DECT Line 1 > Proxy and Registration:**
and then the **Voice tab >DECT Line 1 > Subscriber Information:**

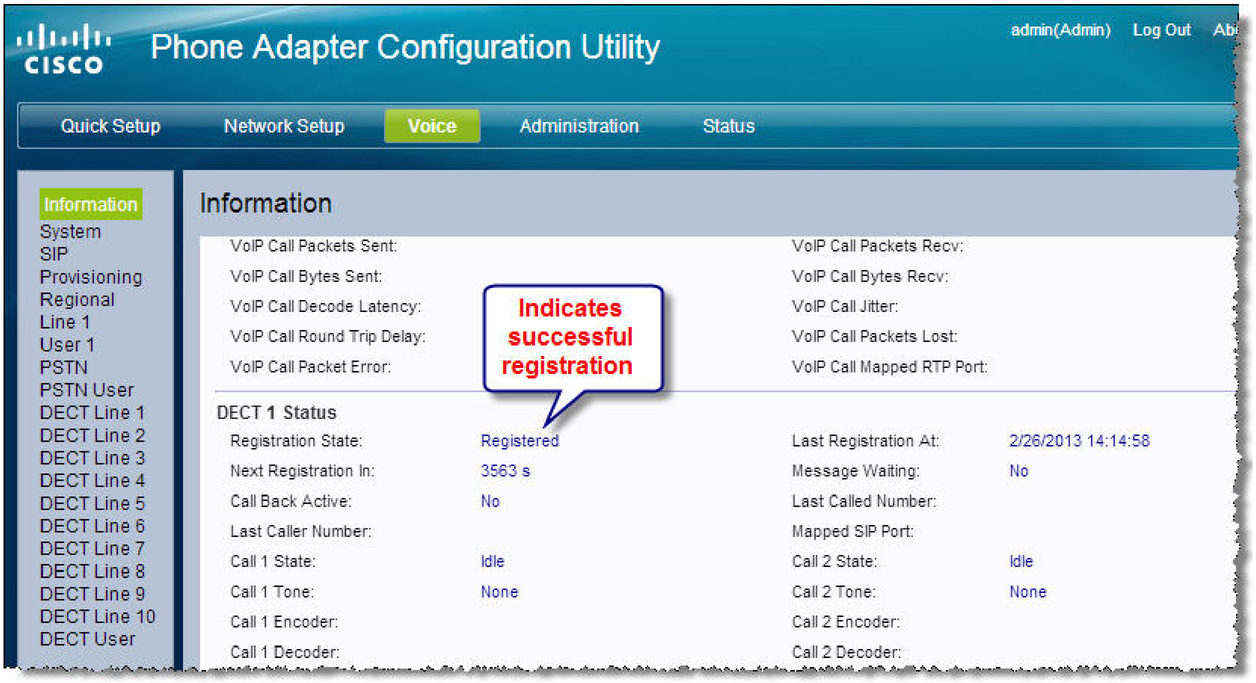


The optional Display Name information is displayed in the SIP Message Header and can be viewed with network packet capture tool such as Wireshark.

Verifying the Registration Status of DECT Line 1

Once you've submitted your ITSP credentials, the SPA232D will restart and attempt to register with the specified ITSP. Verify the registration status with the web-UI at

Voice tab > Information >DECT Line 1 Status:



Day-2 Starts Here

Outbound VoIP Calls

Outbound calls are routed out using VoIP as the default routing because VoIP is usually a lower priced transport. In some cases, you may prefer to force a call to use the PSTN. This section describes using both transport mechanisms.

Configuring Outbound Calls from Analog Phone Connected to Phone Port to use VoIP

Once Line 1 is registered to your ITSP, any inbound calls to the ITSP will make the analog phone or fax connected to the PHONE (FXS) port ring and any outbound calls made from the device connected to the PHONE port of the SPA232D will be routed through the registered ITSP by default.

Using Gateways gw0-4

The SPA232D has 5 internal gateways where:

- gw0—LINE (PSTN / FXO) port of SPA232D
- gw1-4—Outgoing gateways that do not register with a SIP proxy. Usually, the main reason to register to a SIP Proxy is so the proxy knows to what target to send SIP INVITES.
  Configure gw1-4 as follows:
  - Gateway 1-4: userName@houston.voip.ms
  - GW1-4 NAT Mapping Enable: yes
  - GW1-4Auth ID: userName
  - GW1-4 Password: password

Configuring Outbound Calls from Analog Phone Connected to Phone Port to use PSTN Even when Registered to VoIP Account

By default, once Line 1 is registered to a SIP Proxy, all outbound calls are routed out through the SIP Proxy. All outbound calls are routed based on a dial plan. This means that as you dial, the digits are analyzed on the fly and then depending on the dial plan rules, your call is either blocked or routed appropriately. You can use the dial plan to change the way certain calls are routed.

There are multiple dial plans so verify that you are modifying the appropriate dial plan.
Modify the Line 1 Dial Plan located at **Voice tab > Line 1 > Dial Plan > Dial Plan:**

Dialing #9 to Use the PSTN Line for Outbound Call

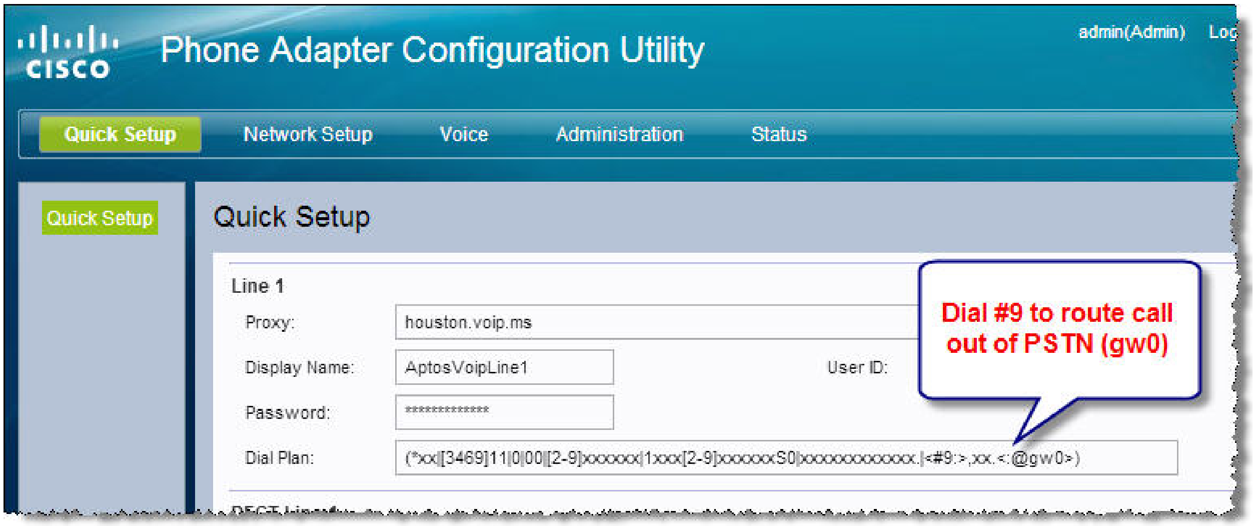
> **WARNING**: Be sure to always leave the emergency dial string near the beginning of the dial plan to ensure that any emergency calls are routed as quickly as possible and are not confused with any dial plan error that you may inadvertently insert into the dial plan. The string containing the emergency number filters is: |**[3469]11**|

Selectively force outbound PSTN routing so that numbers starting with #9 will be routed out of the PSTN by modifying Line 1's default dial plan from:
(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)

To:
(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx. **|<#9:>,xx.<:@gw0>**)

Where:

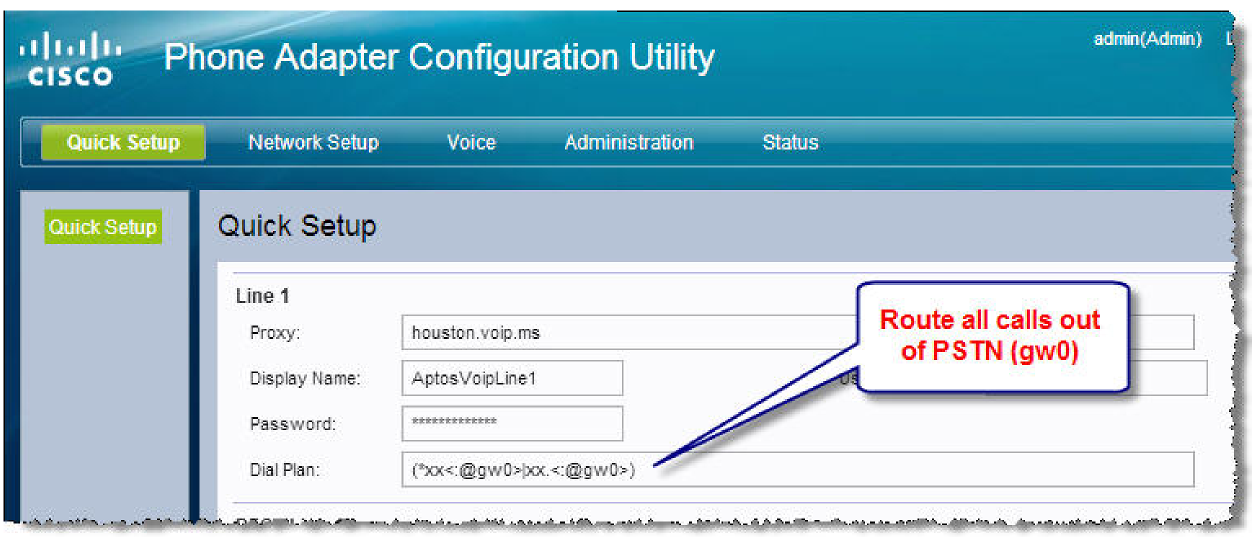
|	—dial string separator
<#9:>	—when hash9 is dialed, discard #9
,	—play dial tone
xx.	—accept one or more digits
<:@gw0>	—route out of gateway 0 (PSTN)



Routing All Line 1 Outbound Calls to use PSTN

Modify dial plan to force all outbound calls out of gw0 (PSTN)

(*xx<:@gw0>|xx.<:@gw0>)



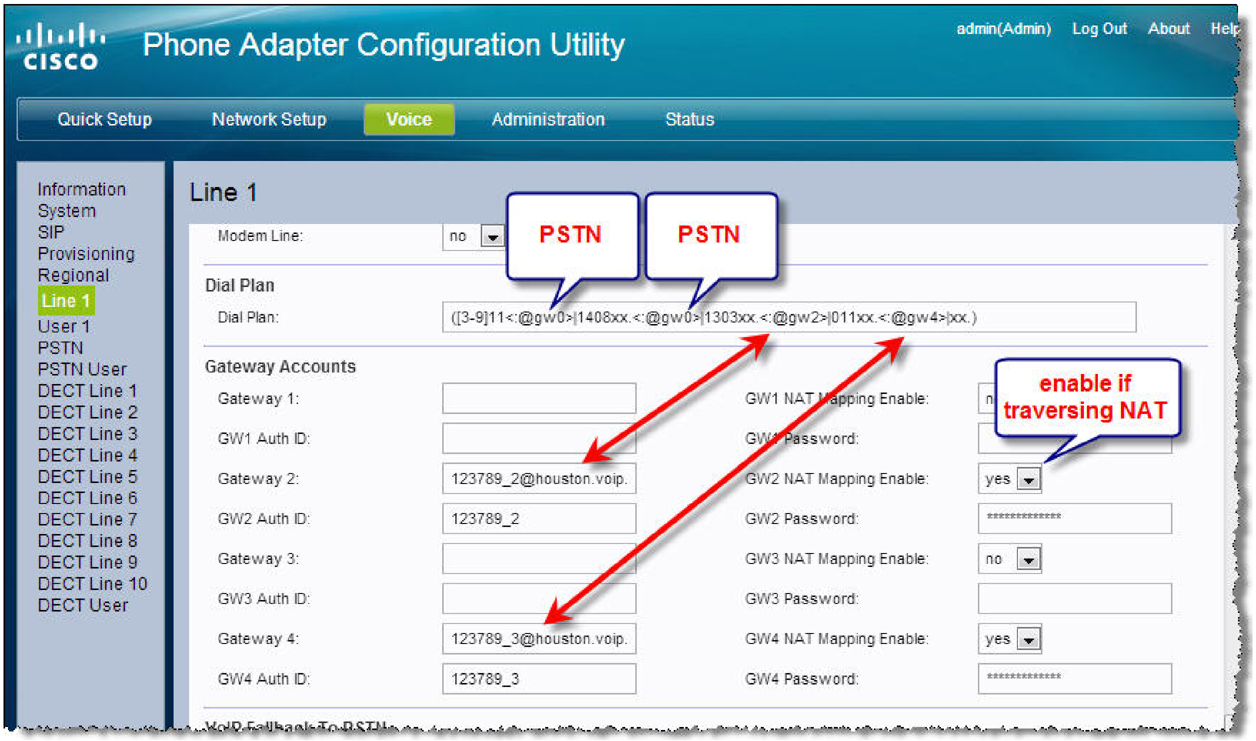
Outbound Routing Using gw0-4 (Scenario)

Scenario: A company has a fax machine connected to the SPA232D and wants to pay the lowest call rates possible. They have different service provider accounts for popular destinations. Analysis shows that they need to filter outbound calls as follows:

- Emergency and similar calls 311, …, 911 must use the PSTN line
- San Jose CA area code 408 must use the PSTN line
- Boulder CO area code 303 must use the SP configured on gw2
- International calls 011 must use the SP configured on gw4
- Any other calls must be routed out of the default ITSP configured on Line 1

Resulting Line 1 Dial Plan:

([3-9]11<:@gw0>|1408xx.<:@gw0>|1303xx.<:@gw2>|011xx.<:@gw4>|xx.)



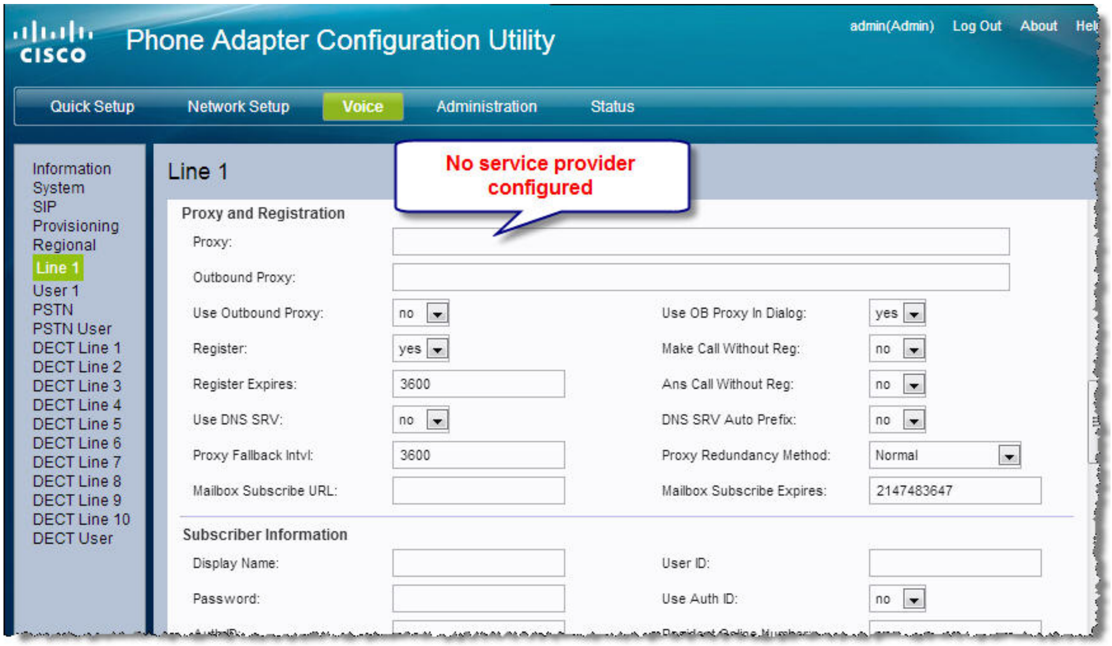
Advanced Call Routing

Aside from the SPA232D's advanced business-grade DECT features, the SPA232D is an advanced ATA and gateway device capable of performing different roles depending on the user's needs.

The DECT base station component of the SPA232D supports up to 5 DECT handsets. Each SPA302D DECT handset can have its own unique DID from an ITSP or share between them. The SPA232D is also capable of being configured as:

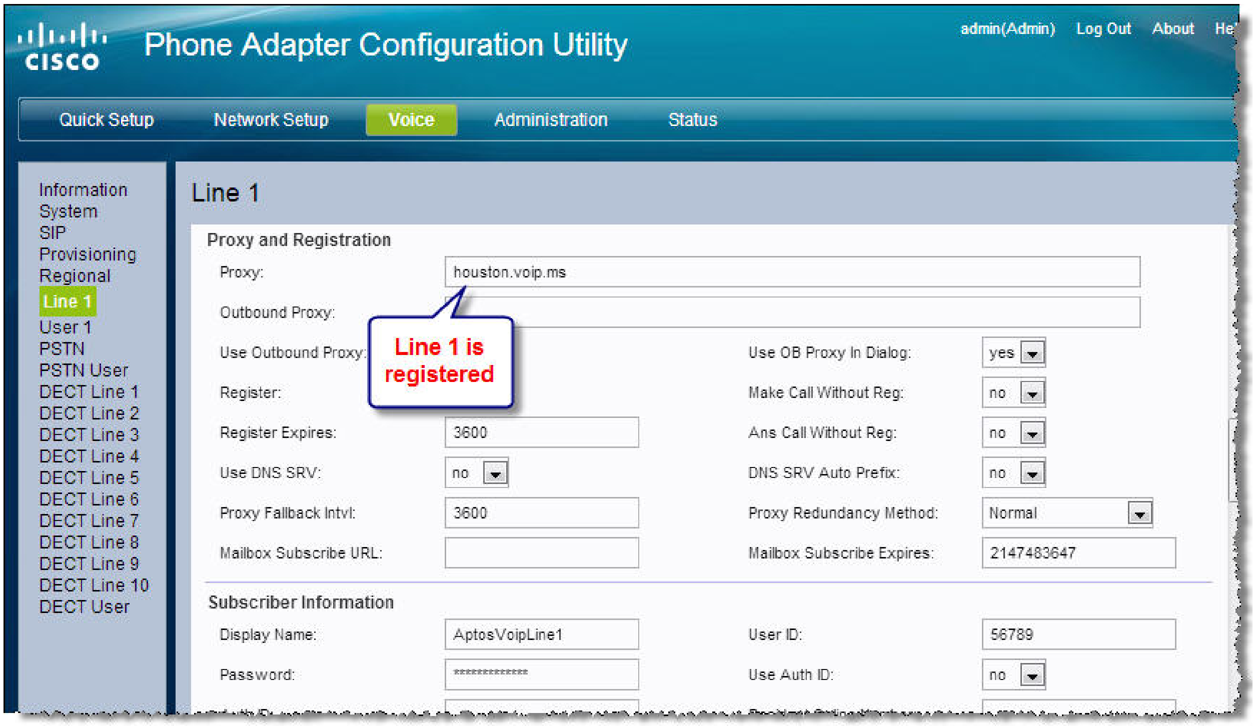
1. A **dumb device**(when not configured in any way) and provides a way to allow an analog phone connected to the PHONE port can make and receive calls via the PSTN connected to the LINE port of the SPA232D even when the SPA232D has no power because its relays are "normally

closed".



Inbound calls will cause the analog phone connected to the PHONE port and all registered SPA302D handsets to ring. The SPA302D handsets can easily be configured to use the PSTN for outgoing calls.

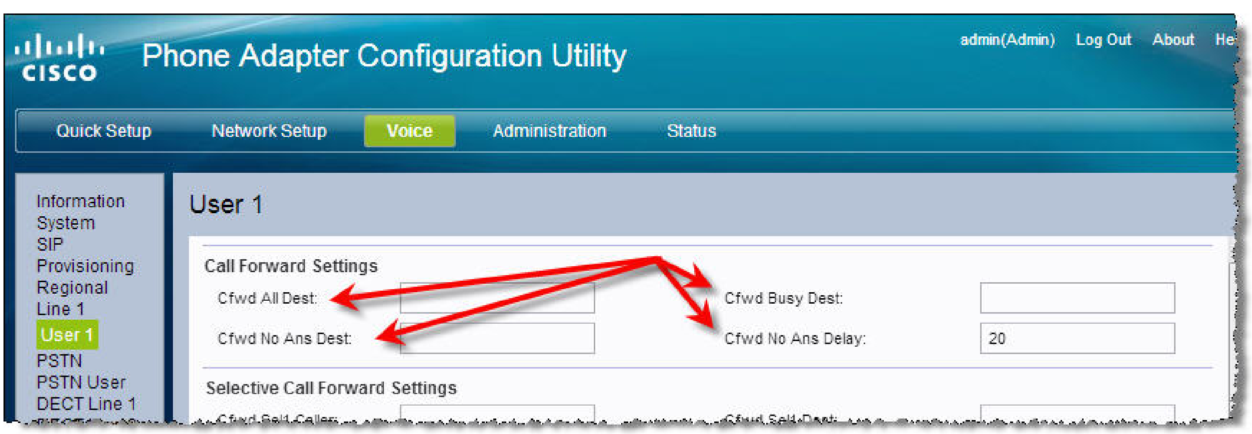
2. An **analog telephone adapter (ATA)**where an analog phone connected to the PHONE port can make and receive calls via the ITSP defined at
Voice tab > Line 1 > Proxy and Registration > Proxy



The analog phone or fax machine connected to the PHONE port of the SPA232D can make and receive calls via the PSTN line connected to the LINE port of the SPA232D or via the configured ITSP. The SPA302D DECT handsets can also make and receive PSTN and ITSP calls.

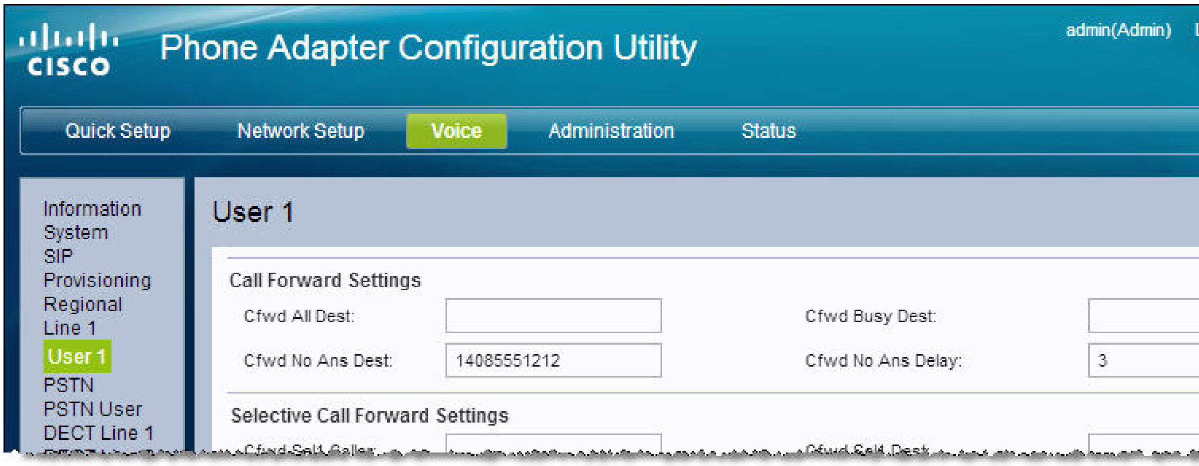
The SPA232D has very powerful call forwarding capabilities where:

- The user can configure call forwarding at **Voice tab > User 1** based on:
  - All calls
  - No answer based on delay timer
  - Busy
  - Based on calling number



- Call forward destinations can be:

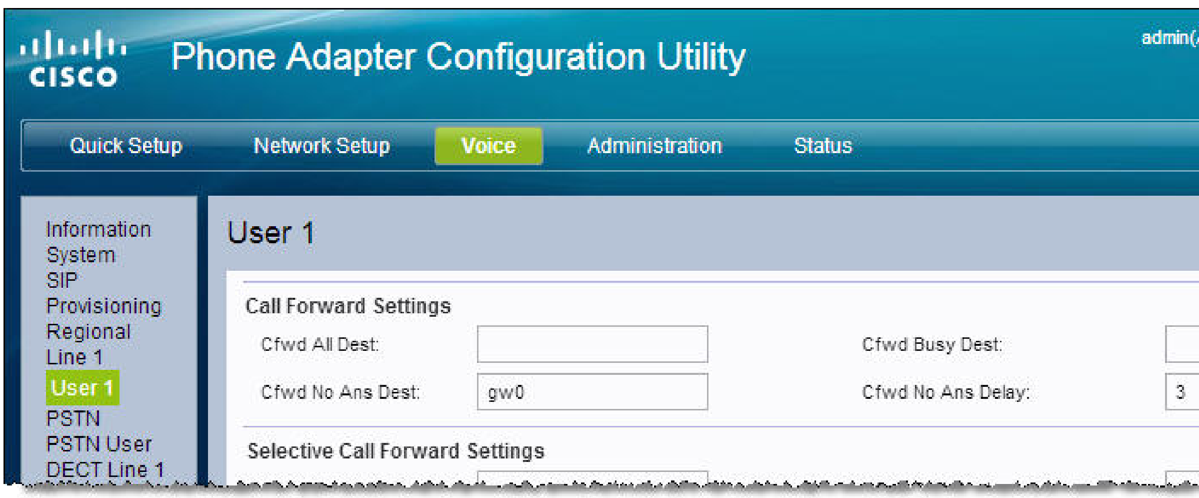
- A telephone number for example: 14085551212



A call to the PSTN number results in a SIP INVITE from the SPA232D to the ITSP registered on Line 1 to the call forward target number.

A call to the ITSP DID registered on Line 1 of the SPA232D results in a PSTN call from the SPA232D to the call forward target number.

- A gateway, for example: gw0



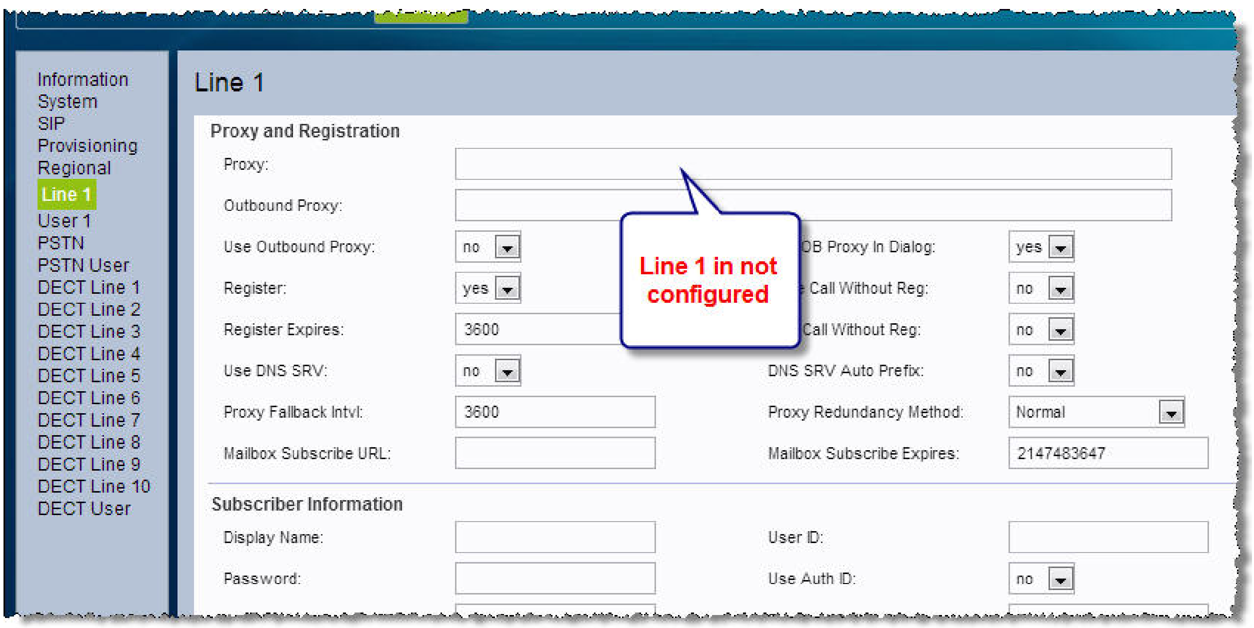
A call to the ITSP DID registered on Line 1 of the SPA232D results in the caller hearing a tone that sounds similar to a long-busy tone. The user is connected to the PSTN connected to the LINE port of the SPA232D and can now dial out of the PSTN and will have the caller ID associated with the PSTN line.

3. A PSTN-to-VoIP gateway where calls in to the PSTN line connected to the LINE port of the SPA232D can dial out using the ITSP configured at the
Voice tab > PSTN > Proxy and Registration

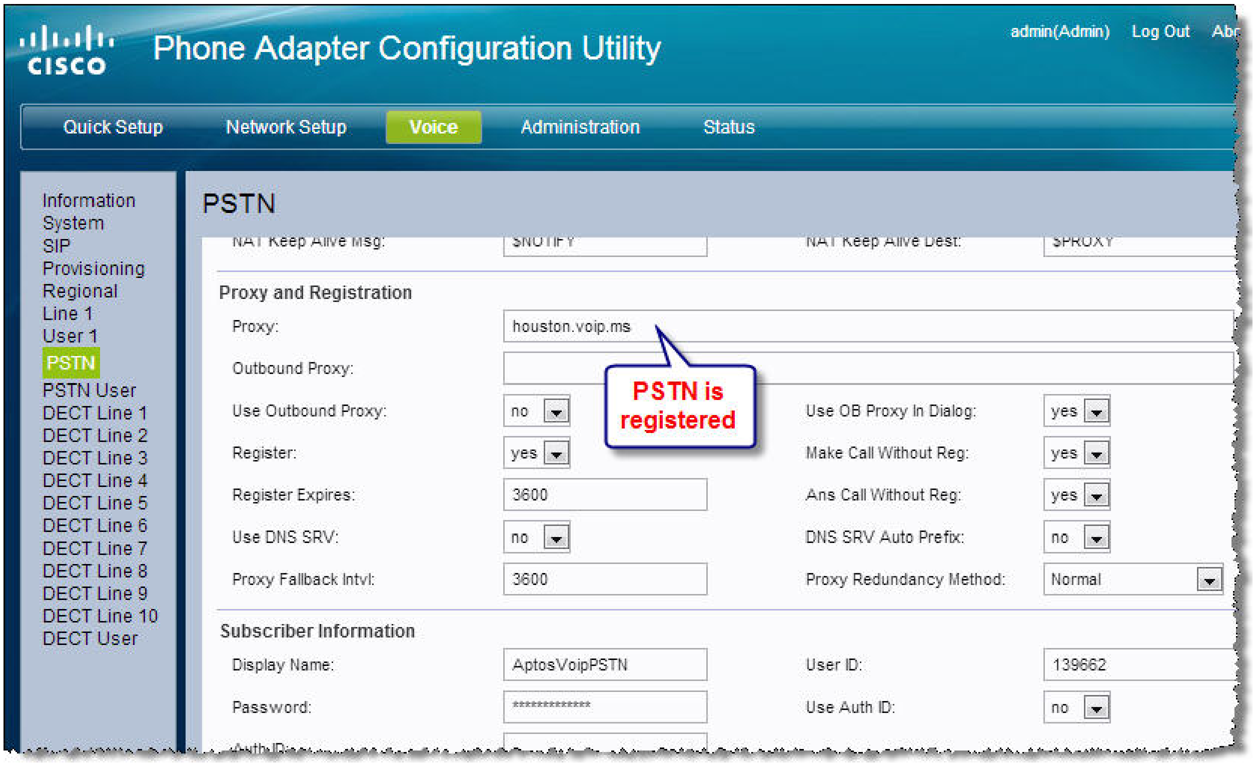
4. A VoIP to PSTN gateway where calls into the ITSP configured at the **Voice tab > PSTN > Proxy and Registration** can dial out using the PSTN line connected to the LINE port of the SPA232D.

   Scenario: A company has international representatives based in other countries. The reps need to make low-cost international calls and appear to be local to the international call recipient. Accomplish this by deploying a SPA232D in the target country and configure with a PSTN and low-cost SP. The rep uses the Internet to access the SPA232D then uses call forward no answer to dial out of the SPA232D's PSTN connection on gw0. The resulting caller ID shows as a local in-country caller. Configure as follows:

   a. Remove the ITSP configuration from Line 1



   b. Configure **Voice tab > PSTN > Proxy and Registration**
   and
   Voice tab > PSTN > Subscriber Information



Inbound calls to the PSTN number causes the analog phone connected to the PHONE port and all registered SPA302D handsets to ring.

A call to the ITSP DID registered on **Voice tab > PSTN** of the SPA232D results in the caller hearing a tone that sounds similar to a long-busy tone. The user is connected to the PSTN connected to the LINE port of the SPA232D and can now dial out of the PSTN and will have the caller ID associated with the PSTN line. Contrast this direct-connect behavior with the PSTN line as compared to where the call had to be forwarded to the gw0 gateway when Line 1 was used.

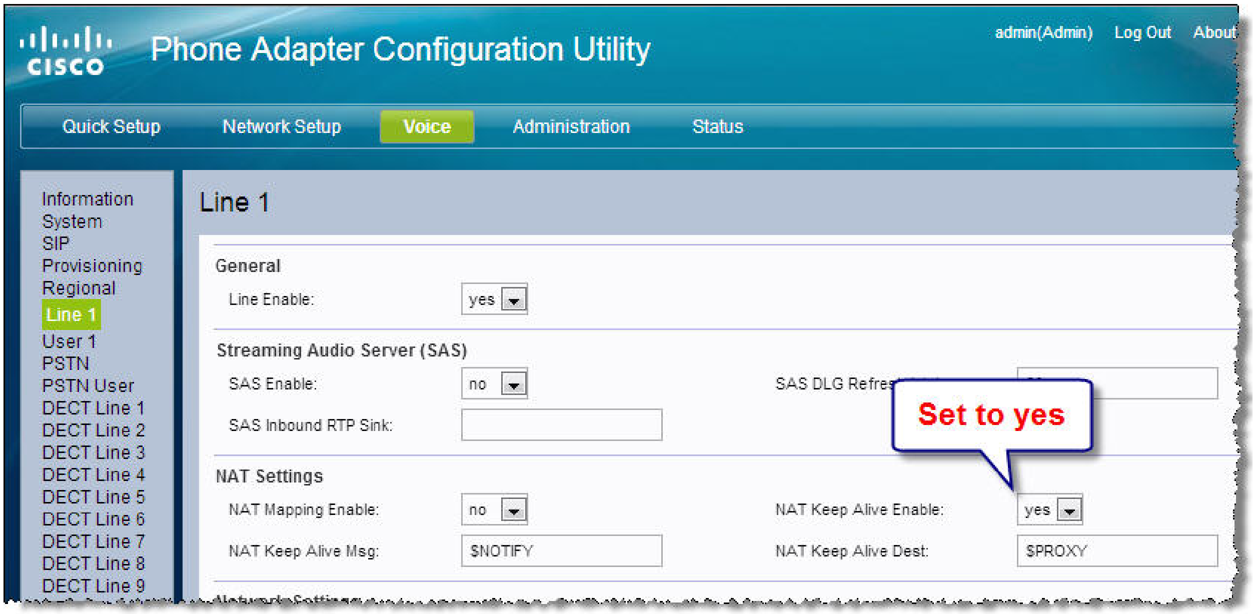
5. You can also configure authentication for the VoIP to PSTN gateway. The https://supportforums.cisco.com/docs/DOC-9902document although written for the SPA3102 remains current for this task on the SPA232D

Troubleshooting:

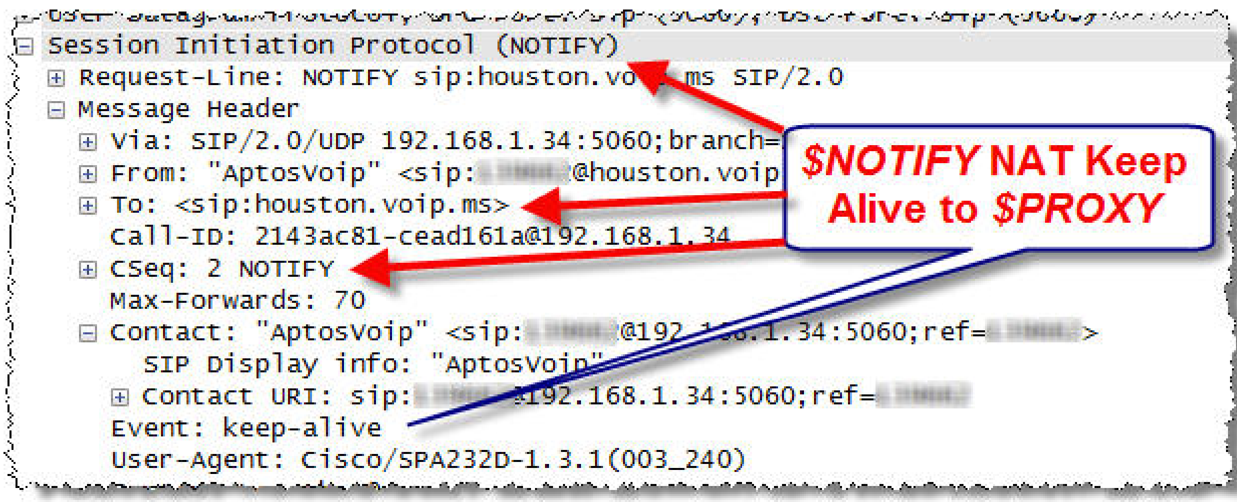
Calls to ITSP Intermittently Result in a Busy Signal

Enable the NAT Keep Alive feature for the problematic connection as follows:

- For Line 1: **Voice tab > Line 1> NAT Settings > NAT Keep Alive Enable:** yes
- For PSTN: **Voice tab > PSTN > NAT Settings > NAT Keep Alive Enable:** yes
- For DECT Line *N*: **Voice tab > DECT Line *N*> NAT Settings > NAT Keep Alive Enable:** yes



The SPA232D will now send a SIP NOTIFY message to the destination defined by resolving the $PROXY macro:



Additional Resources

Quick Start Guides:

- Cisco SPA232D Mobility Enhanced ATA Quick Start Guide
- Cisco SPA302D Mobility Enhanced Cordless Handset Quick Start Guide

User Guide:

- SPA232D does not have a User Guide
- Cisco Small Business SPA302D Mobility Enhanced Cordless Handset User Guide

Administration Guide:

- Cisco SPA232D Mobility Enhanced Phone Adapter Administration Guide
- SPA302D does not have a separate Administration Guide.
  All configuration is performed on the SPA232D

Provisioning Guide:

- Provisioning Guide for SPA112, SPA122, and SPA232D Analog Telephone Adapters
- The SPA302D does not have a separate Provisioning Guide.
  All provisioning is performed for the SPA232D

Software Release Notes:

- Cisco SPA232D Mulit-Line DECT ATA Software Release Notes
- The SPA302D does not have separate Software Release Notes.
  The SPA302D firmware is included in the SPA232D's firmware

Firmware / Software

- Cisco SPA232D Multi-Line DECT ATA Software
- The SPA302D firmware is included in the SPA232D firmware

<end>