Application Note

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Configuring the Cisco SPA ATA Family with Skype for SIP

Cisco SPA8000 8-Port IP Telephony Gateway-Specific Configuration

Check for updates of this document at:
https://www.myciscocommunity.com/docs/DOC-ToBeAssigned

Introduction

This document assumes that you have a Cisco Smart Phone Appliance (SPA) Analog Telephony Adapter (ATA), [SPA ATA] that you want to configure to use Skype for SIP. This means that you will be able to use a Skype for SIP account and be able to make and receive phone calls with regular analog phones connected to the ATA.

This document uses the SPA8000 8-Port IP Telephony Gateway in all examples. The SPA8000 supports up to four trunk groups, numbered T1, T2, T3, and T4. Skype for SIP can be configured on each trunk group with a distinct phone number. Each of the eight SPA8000 lines can be configured either as a standalone line, as in a classic ATA FXS port, or as a trunk line that is associated with a trunk group.

- Inbound calling: A trunk group offers a single number for callers to call into the small business, with the capability to programmatically ring one or more trunk lines.
- Outbound calling: When a PBX phone makes a call, the PBX selects one of the available trunk lines. The trunk line assumes the Caller ID of the trunk group.

Both trunk-mode and classic ATA FXS port-mode are configured in this application note.

The following table provides a summary of the Sipura > Linksys > Cisco SPA ATA family features:

Model	Service Lines (FXS)	PSTN Port (FXO)	PC Port(s) (10/100)	Active Calls	3-Way Calls	USB Port
PAP2T	2	0	N	2	1	N
SPA2102	2	0	1	4	2	N
SPA3102	2	1	1	3	1	N
SPA8000	8	0	N	16	8	N
SPA8800	4	4	N	12	4	N
WRP400	2	0	4	4	2	Y

Audience

This application note is targeted to anyone with a SPA ATA and analog phones who wants to save money on phone calls by leveraging Voice over IP (VoIP) and Skype for SIP.

Scope

This scope of this document is limited to configuring inbound and outbound call routes on a SPA ATA and analog phones and does not address the following topics:

- Security
- Acquiring a Skype for SIP account
- Advanced call routing

Refer to the Related Documents for additional configuration and background information.

Related Documents & Resources

- SPA ATA family data sheets
- SPA ATA Administration Guide
- Cisco Community Central: ATAs, Gateways and Accessories
- Skype for SIP

Overview

For the purposes of this document, the SPA8000 8-Port IP Telephony Gateway is configured with Skype for SIP. The SPA8000 will first be configured in trunk-mode and then in classic ATA FXS port-mode.

By the end of this document, you will be able to connect and configure a SPA ATA to a previously configured Skype for SIP account and make low cost calls using regular analog telephones.

Summary of Tasks in this Document

You must complete the following tasks in order to configure Skype for SIP with a SPA9000 Voice System:

1. Procure Skype for SIP credentials. Refer to the following Skype site for subscription information and additional information about Skype for SIP:
   http://www.skype.com/business/products/pbx-systems/sip/
   Following are example credentials:

SIP User:	99051000000420
Password:	a6dVfzgvMM7xyz
Skype for SIP address:	sip.skype.com
UDP Port:	5060
STUN address:	stun.skype.com

2. Factory reset the SPA8000 so that you can configure from a known state.
3. Decide for which mode the SPA8000 must be configured
   a. Skype for SIP trunk-mode

   b. Skype for SIP classic ATA FXS port-mode

4. Configure the SPA8000 to register to the Skype for SIP service.
5. Save changes which causes the SPA8000 to reboot and register to Skype for SIP

Requirements

You need the following equipment, services, and information:

- Skype for SIP credentials
- At least one phone number [Direct inward dial (DID)] from Skype for SIP
- A working Internet connection

Configuring the SPA8000 for Skype for SIP

In this document two SPA8000 configuration examples are provided, classic ATA FXS port-mode and trunk group-mode.

You only need to follow one configuration example, depending on the configuration you prefer. The beginning of each example describes the characteristics of the feature.

1. Factory Resetting the SPA8000

Factory reset the SPA8000 so that you can configure from a known configuration:

a. Connect an analog phone to Phone line connection 1 or 2 on the SPA8000.
b. Connect the SPA8000's ETHERNET port to the LAN so that it can access a DHCP server for a dynamic IP address and access the Internet.
c. Power on the SPA8000.
d. After about 60-seconds, go off-hook on the analog phone, you will not hear anything on the handset.
e. Dial **** (four stars/asterisks) and you will hear the configuration menu.
f. Dial **RESET#** (73738#) to factory reset the SPA8000.
g. Dial 1 to confirm the factory reset and the SPA8000 will reboot.

2. i. Configuring the SPA8000 in Classic ATA FXS-Mode [Optional]

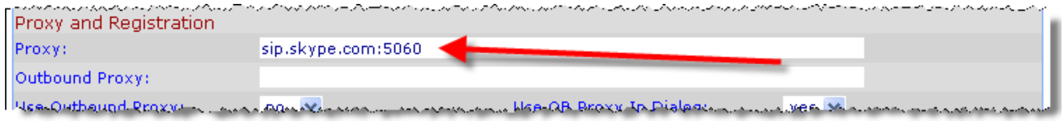
The SPA8000 must have at least one phone line or trunk group registered to the Skype for SIP service in order to make and receive calls.

When an analog phone makes a call, the SPA8000 uses the Skype for SIP account registered to the phone line. The phone line assumes the Caller ID of the Skype for SIP account to which it is registered.

Configure line 1 in classic ATA FXS-mode as follows:

a. Access the SPA8000's web user interface (web-UI)
   http://<SPA8000_IP_address>/admin/advanced

b. Insert **`sip.skype.com:5060`** into the field at:
   SPA8000 Voice tab > L1 tab > Proxy and Registration > Proxy:

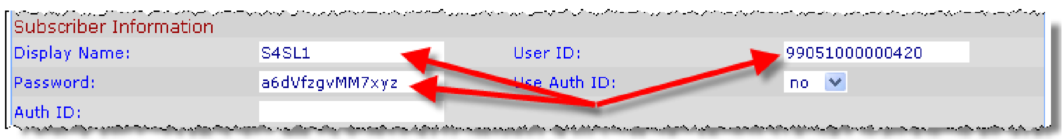


c. Insert the Skype for SIP user credentials in the
   SPA8000 Voice tab > L1 tab > Subscriber Information >

   a. Display Name: [Any name to help you identify this line in a diagnostic trace]

   b. User ID: [***SIP User*** from Skype for SIP credentials]

   c. Password: [***Password*** from Skype for SIP credentials]



a. Configure additional SPA8000 lines with other Skype for SIP accounts as needed.

b. Click Submit All Changes. The SPA8000 will reboot and be ready for use.

This completes configuring the SPA8000 to interoperate with Skype for SIP in classic ATA FXS port mode.

2. ii. Configuring the SPA8000 in Trunk-Mode *[Optional]*

The SPA8000 must have at least one trunk group or one phone line registered to the Skype for SIP service in order to make and receive calls.

The SPA8000 supports up to four trunk groups, numbered T1, T2, T3, and T4. Each trunk group can be registered to a Skype for SIP account providing a distinct phone number for each trunk group.

A trunk group offers a single number for callers to call into the small business, with the capability to programmatically ring one or more phone lines associated with the trunk group.

When an analog phone makes a call, the SPA8000 selects one of the available trunk lines. The trunk line assumes the Caller ID of the trunk group.

Configure the SPA8000's T1 trunk to register to the Skype for SIP service as follows:

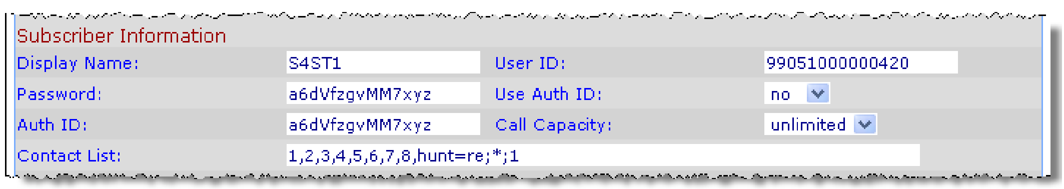
a. Access the SPA8000's web user interface (web-UI)
   http://<SPA8000_IP_address>/admin/advanced

b. Insert the Skype for SIP user credentials in the
   SPA8000 Voice tab > T1 tab > Subscriber Information >

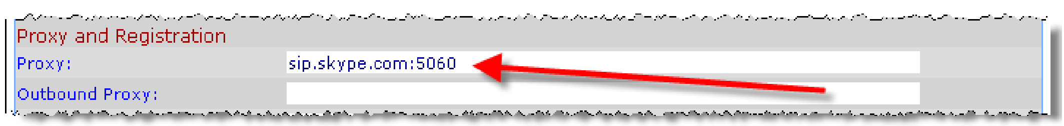
   a. Display Name: [Any name to help you identify this line in a diagnostic trace]

   b. User ID: [***SIP User*** from Skype for SIP credentials]

   c. Password: [***Password*** from Skype for SIP credentials]



c. Insert **`sip.skype.com:5060`** into the field at:
   SPA8000 Voice tab > T1 tab > Proxy and Registration > Proxy:



d. Configure additional SPA8000 lines with other Skype for SIP accounts as needed.

e. Click Submit All Changes. The SPA8000 will reboot and be ready for use.

This completes configuring the SPA8000 to interoperate with Skype for SIP in trunk-mode.

3. *Testing*

i. Test outbound calls by making external calls to:

a. Local 7-digit number

b. Local 10-digit numbers

c. Long distance 11-digit numbers

d. International calls

ii. Test inbound call routing by making a call to the Skype for SIP DID

Troubleshooting

There are multiple reasons for calls to fail. The most efficient way to troubleshoot calling problems is to break down the issue in to either outbound or inbound call problems.

Troubleshooting: One-way Audio

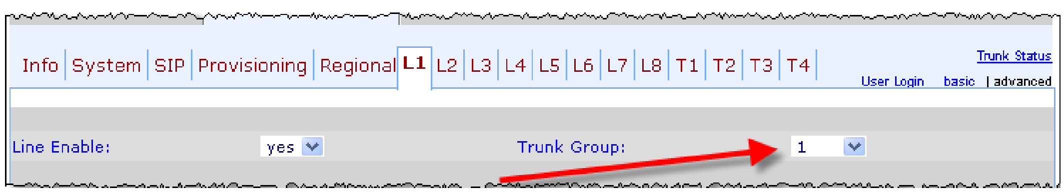
If you call someone and they can hear you, but you cannot hear them, or vice-versa, you are experiencing one-way audio.

One way audio is a symptom of missing voice data. Voice data may get lost in the Internet if your device is located behind a network address translator (NAT). When a VoIP conversation is initiated, all initiation is performed by SIP. As soon as voice traffic is about to flow, the Real-Time Protocol (RTP) stream is started. SIP takes the long way through the Internet, following all routes until the destination is located. Because voice traffic is time sensitive, RTP takes a direct route. This sometimes results in problems. This is why Skype make a STUN server available.

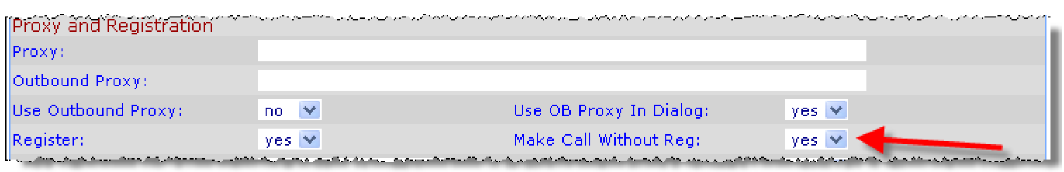
Refer to the Configuring STUN [Optional] section for more information and configuration instructions.

Troubleshooting: LED for Phone is off [Skype for SIP Trunk]

The LED representing the phone attached to Phone *N* on the SPA8000 only illuminates when the phone line is registered to either a T*N* trunk or to a SIP proxy such as Skype for SIP and the line is configured to make a call without being registered
SPA8000 web-UI > Voice tab > Line*N* > Trunk Group: 1-4



SPA8000 web-UI > Voice tab > Line*N* > Proxy and Registration > Make Call Without Reg:

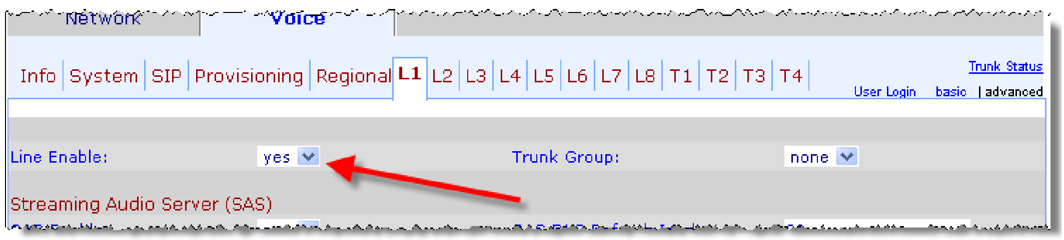


Troubleshooting: *LED for Phone is off [ATA FXS Port Mode]*

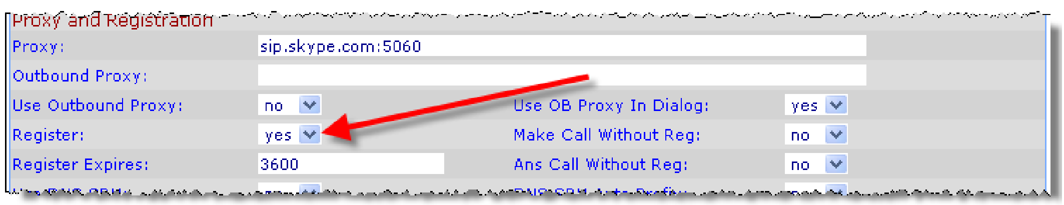
This symptom is usually due to the incorrect user credentials being used, such as an incorrect password, the line not being enabled, or the line not being configured to register with Skype for SIP.

1. Verify that the line is registered:
   SPA8000 web-UI > Voice tab > Info tab > Line *N* Status > Registration State:

2. Verify that the line is enabled:
   SPA8000 web-UI > Voice tab > L*N* tab > Line Enable:



3. Verify that the line is configured to register to Skype for SIP:
   SPA8000 web-UI > Voice tab > L*N* tab > Proxy and Registration > Register:



Troubleshooting: *Outbound Call Rings Once Then Busy*

This symptom is usually due to dialing a number that is valid for the SPA8000 but not valid for Skype for SIP. An example is dialing a long distance number in the USA without dialing a 1. A valid long distance number is 1-NNN-NNN-NNNN. Dialing NNN-NNN-NNNN is valid for the SPA8000's dial plan, but Skype for SIP requires the 1 to be dialed for long distance numbers.

Troubleshooting: *Inbound Calls with a Skype for SIP Trunk*

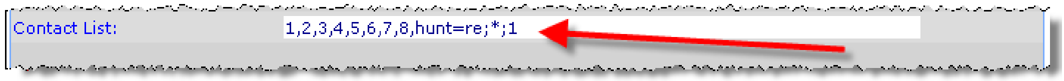
Inbound caller hears busy.

1. Verify that the SPA8000's trunk is registered:
   SPA8000 web-UI > Voice tab > Info tab > Trunk *N* Status > Registration State:

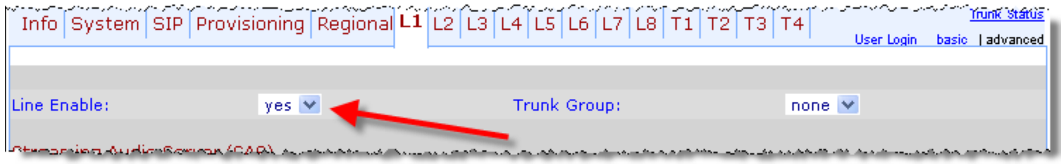
2. Verify that the inbound call has a valid call route:

   a. Verify that the trunk's contact list includes at least one SPA8000 Phone line to which an analog phone is connected. The default contact list will ring all

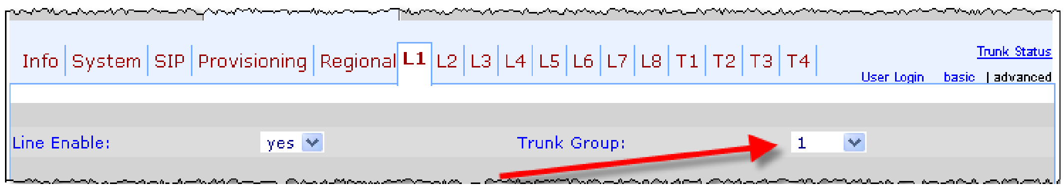
connected analog phones starting at phone 1 then phone 2 and so on:
SPA8000 web-UI > Voice tab > T*N* tab > Subscriber Information > Contact List:



b. Verify that the relevant line is enabled:
SPA8000 web-UI > Voice tab > Line*N* > Line Enable: yes



c. Verify that the relevant line is associated with a Trunk Group:
SPA8000 web-UI > Voice tab > Line*N* > Trunk Group: 1-4

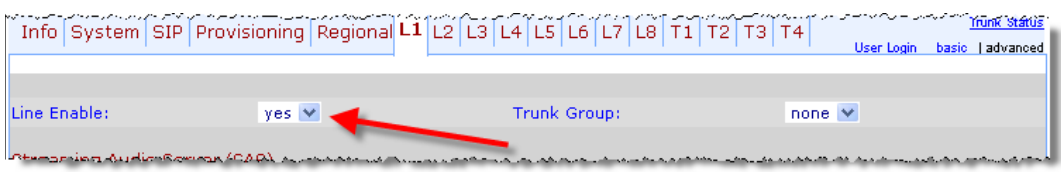


If after verifying all of the above steps, the inbound caller still hears busy, consider factory defaulting the SPA8000 and configuring it again by performing only the steps listed in this application note. The SPA8000 has many configurable parameters, changing any one of the parameters without full understanding of the full impact of the change can result in inbound call failure.

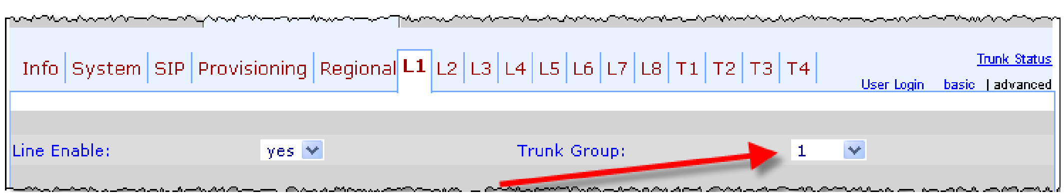
Troubleshooting: Outbound Calls with a Skype for SIP Trunk

Outbound caller hears busy.

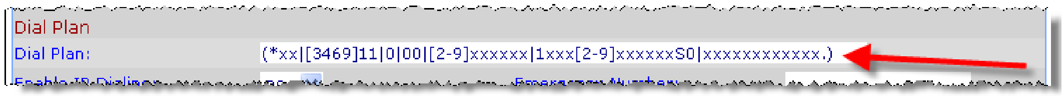
1. Verify that the relevant line is enabled:
SPA8000 web-UI > Voice tab > Line*N* > Line Enable: yes



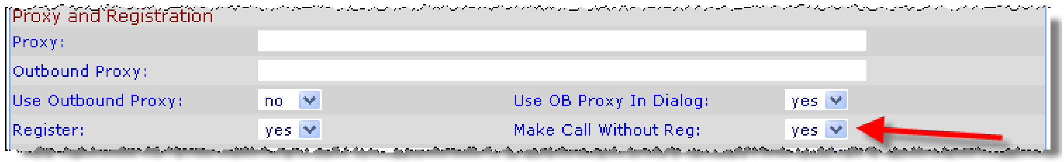
2. Verify that the relevant line is associated with a Trunk Group:
SPA8000 web-UI > Voice tab > Line*N* > Trunk Group: 1-4



3. Verify that the SPA8000's phone line's dial plan allows the dialed number. The default dial plan allows all numbers to be dialed:
   SPA8000 web-UI > Voice tab > Line*N* >



4. Verify that the SPA8000's line allows calls to be made with a trunk, that is making calls without being registered:
   SPA8000 web-UI > Voice tab > Line*N* > Proxy and Registration > Make Call Without Reg:



5. Verify that the SPA8000's line is registered to the SPA8000 trunk:
   SPA8000 web-UI > Voice tab > Info tab > Line *N* Status > Registration State:

6. Verify that the SPA8000's trunk is registered:
   SPA8000 web-UI > Voice tab > Info tab > Trunk *N* Status > Registration State:

If after verifying all of the above steps, the outbound caller still hears busy, consider factory defaulting the SPA8000 and configuring it again by performing only the steps listed in this application note. The SPA8000 has many configurable parameters, changing any one of the parameters without full understanding of the full impact of the change can result in outbound call failure.

Configuring STUN [Optional]

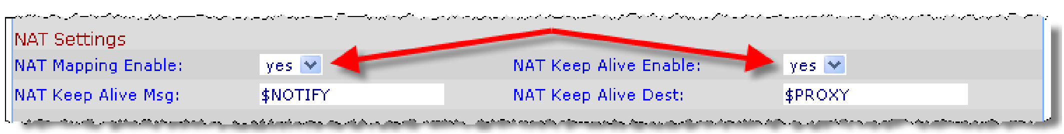
Skype for SIP user account credentials include STUN information. You may need to use STUN if your connection to the Internet has an IP address from a network address translator (NAT).

The **S**imple **T**raversal **U**tilities for **N**AT (STUN) [defined in RFC5389] provides a way for the SPA8000 to make VoIP phone calls with SIP when your network devices do not have a static IP address and port associated with them.

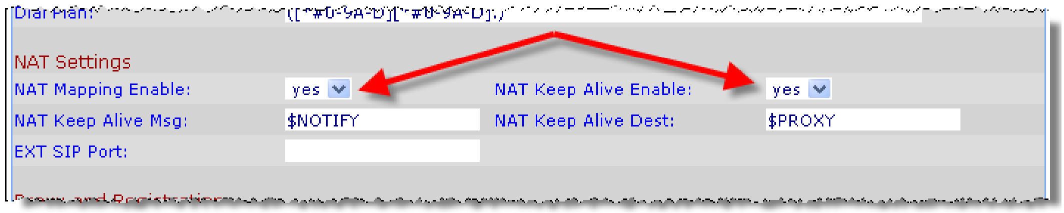
Configure STUN as follows:

1. Enable NAT support parameters and STUN
   SPA8000 web-UI > Voice tab > SIP tab > NAT Support Parameters >
    a. Handle VIA received: yes
    b. Handle VIA rport: yes
    c. Insert VIA received: yes
    d. Insert VIA rport: yes
    e. Substitute VIA Addr: yes
    f. Send Resp to Src Port: yes
    g. STUN Enable: yes
    h. STUN Server: stun.skype.com

2. i. [Optional: **only if you configured classic ATA FXS port-mode**] Enable NAT
   SPA8000 web-UI > Voice tab > L*N* > NAT Settings >
    a. NAT Mapping Enable: yes
    b. NAT Keep Alive Enable: yes



2. ii. [Optional: **only if you configured Trunk-mode**] Enable NAT
   SPA8000 web-UI > Voice tab > T*N* > NAT Settings >
    a. NAT Mapping Enable: yes
    b. NAT Keep Alive Enable: yes



3. Click Submit All Changes to save and reboot the SPA8000.

Gathering Information for Support

In the event that you need to reach out for support, collect the following information first:

A. SPA 8000's configuration: https://www.myciscocommunity.com/docs/DOC-10116

B. ATA Documentation: https://www.myciscocommunity.com/docs/DOC-2149

C. WireShark trace to allow the support staff to view network interaction.