Configuring the Cisco SPA9000 IP PBX with a Skype for SIP Trunk:

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Cisco Small Business Pro

Introduction

This document assumes that you have a configured SPA9000 voice system and that you are interested in adding a Skype for SIP trunk to the SPA9000. This means that you will be able to use a single Skype for SIP account and be able to make multiple simultaneous outbound phone calls with analog and IP phones that are registered to the SPA9000 IP PBX.

Audience

This application note is targeted to any SPA9000 administrator who wants to save money on phone calls by leveraging the SPA9000 Voice System with Skype for SIP.

Scope

This scope of this document is limited to configuring inbound and outbound call routes on a SPA9000 and does not address the following topics:

- Installing a SPA9000 Voice System
- Configuring a SPA9000 Voice System, aside from the Skype for SIP trunk
- Security
- Acquiring a Skype for SIP account

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

Related Documents & Resources

- SPA9000 Datasheet
- SPA9000 Voice System Administration Guide
- Installation and Configuration of the SPA9000 Voice System [with the Wizard]
- Installation and Configuration of the SPA9000 Voice System [with the web-UI]
- SPA9000 Setup Wizard
- Cisco SPA500 User Guide
- Cisco SPA500 Series and WIP310 IP Phone Administration Guide
- Cisco Community Central: Small Business Voice Systems-formerly LVS SPA9000
- Cisco Community Central: Small Business Community IP Phone Support
- Skype for SIP

Overview

For the purposes of this document, the SPA525G 5-line IP phone with wideband audio support, Bluetooth, and color screen is used in most examples.

By the end of this document, you will be able to connect and configure a SPA9000 to a previously configured Skype for SIP account and make low cost calls.

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. Configure a line on the SPA9000 for Skype for SIP. Line 2 in this document.
3. Configure the SPA9000's line's dial plan.
4. Configure the SPA9000's call routing rule in order to route outbound calls to the Skype for SIP trunk
5. Configure the SPA9000's phone dial plan to cause the SPA9000 to automatically update all IP phones with the appropriate dial plan
6. Save changes which causes the SPA9000 to reboot, update all IP phones and register to Skype for SIP

Requirements

You need the following equipment, services, and information:

- A functional SPA9000 Voice System
- Skype for SIP credentials
- Optional for receiving calls from Skype for SIP:
  At least one phone number [Direct inward dial (DID)] from Skype for SIP
- A working Internet connection

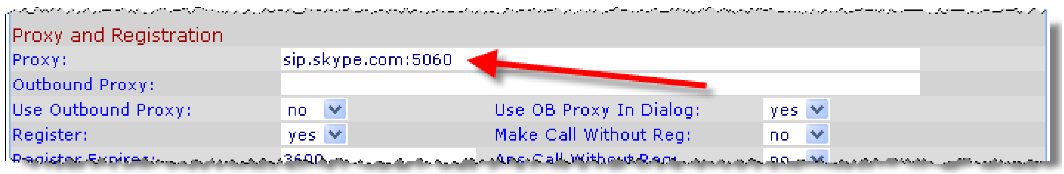
Configuring the SPA9000 for Skype for SIP

In this document's example:

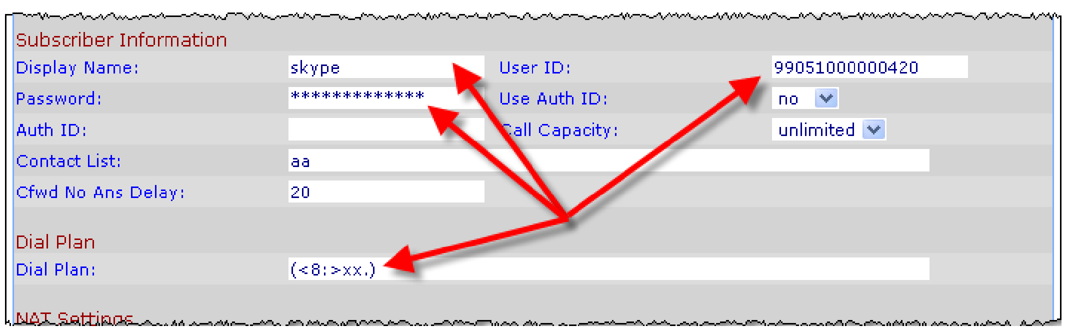
- SPA9000 line 2 is used for Skype for SIP
- 8 is to be the steering digit [SPA9000 currently uses 9 for SPA400 PSTN calls]
- Inbound calls are routed to the auto attendant (aa)

Configure the SPA9000's line 2 to register to the Skype for SIP service:

1. Access the SPA9000's web user interface (web-UI)
   `http://<SPA9000_IP_address>/admin/advanced`
2. Insert **`sip.skype.com:5060`** into the field at:
   SPA9000 Voice tab > Line 2 tab > Proxy and Registration > Proxy:



3. Insert the Skype for SIP user credentials in the
   SPA9000 Voice tab > Line 2 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]
4. Edit the dial plan so that line 2 uses 8 as a steering digit as follows: SPA9000 Voice tab > Line 2 tab > Dial Plan > Dial Plan: **`(<8:>xx.)`**

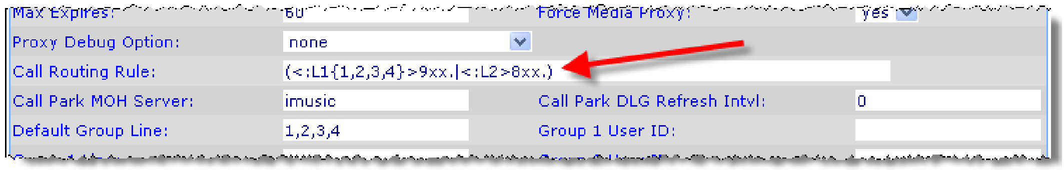


Configuring the SPA9000's Call Routing Rule

Edit the SPA9000's call routing rule so that the SPA9000 routes calls prefixed with a steering digit of 8 to line 2 of the SPA9000. Configure the call routing rule as follows:

1. Locate the existing call routing rule for line 2 at:
   SPA9000 web-ui > Voice tab > SIP tab > PBX Parameters > Call Routing Rule:
   `(<:L1{1,2,3,4}>9xx.)`

2. Modify the call routing rule to: (<:L1{1,2,3,4}>9xx.|**<:L2>8xx.**)

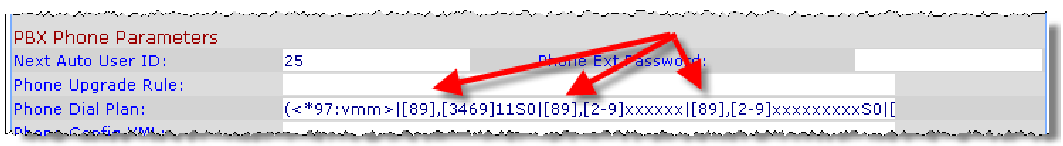


Configuring the SPA9000's Phone Dial Plan

Configure the phone dial plan on the SPA9000 so that the SPA9000 can automatically provision IP phones with the appropriate dial plan.

1. Locate the existing phone dial plan at:
   SPA9000 web-UI > Voice tab > PBS Phone Parameters > Phone Dial Plan:
   (<*97:vmm>|9,[3469]11S0|9,[2-9]xxxxxx|9,[2-9]xxxxxxxxxS0|
   9,1[2-9]xxxxxxxxxS0|9,011xx.|[12345678]|[12345678]x)

2. Modify the phone dial plan by adding an 8 to the steering digit for each sequence and removing the two 8s from the allowed dialing sequence [colored red in step 1]:
   (<*97:vmm>|[**8**9],[3469]11S0|[**8**9],[2-9]xxxxxx|[**8**9],[2-9]xxxxxxxxxS0|
   [**8**9],1[2-9]xxxxxxxxxS0|[**8**9],011xx.|[1234567]|[1234567]x)



3. Click Submit All Changes. The SPA9000 will reboot and cause all the IP phones to reboot.

This completes configuring the SPA9000 to interoperate with Skype for SIP.

Testing

1. 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
2. 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 Dial Plans [Outbound calls]

Dial plans can be difficult to troubleshoot. Consider starting with a simple string and testing before defining a complex dial plan.

Outbound calls can fail if the phone's dial plan is too strict and fails to allow the dialed number to be passed on. This results in a busy tone. Determine if the problem is related to the steering digit or to the subsequent numbers as follows:

1. Go off-hook and hear the phone play dial tone: "pooooooooo …"
2. Dial the steering digit and the phone should play the outbound dial tone: "paaaaaaaa …"

   If the phone plays a busy signal, this indicates that the phone's dial plan does not permit the steering digit. Review the phone's dial plan for 8, in the appropriate places. Refer to the phone's administration guide.

   If the phone does not make a sound after the steering digit is dialed, this indicates that the phone allows the dialed digit, but does not recognize it as a steering digit. Review the phone's dial plan after the 8**,**. A common error is to accidentally configure an internal dial sequence that includes the steering digit. For example, …|[1-**8**]xx.|… would be an error because 8 is a steering digit and must not be used for internal dialing. Correct the string by changing it to …|[1-7]xx.|… by removing the 8. Save changes and test again.

Troubleshooting Registration

Registration issues can result in both inbound and outbound call failure. Registration will fail if the network has a problem or if incorrect user credentials are used, for example, incorrectly typing a password. Verify that the SPA9000's Skype for SIP line is registered as follows:

SPA9000 web-UI > Info tab > Line 2 Status > Registration State:

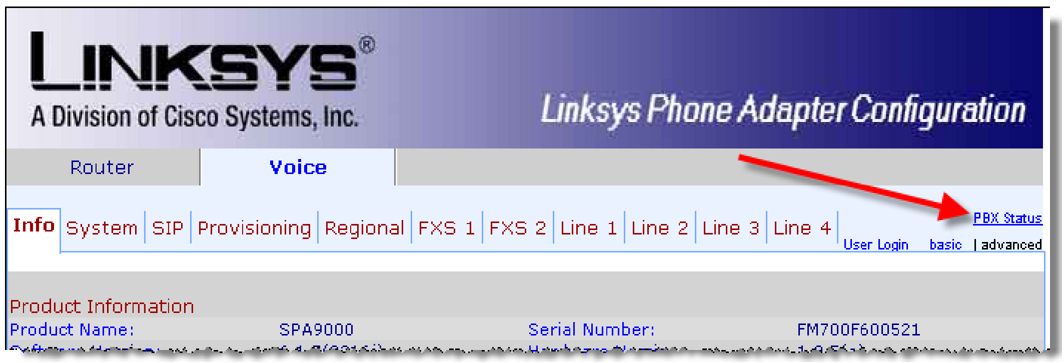
Troubleshooting Inbound Calls

Inbound calls that hear busy are usually a result of the SPA9000 Line's Contact List being incorrect. Verify that there are no syntax errors in the SPA9000 Voice tab > Line 2 tab > Subscriber Information > Contact List:

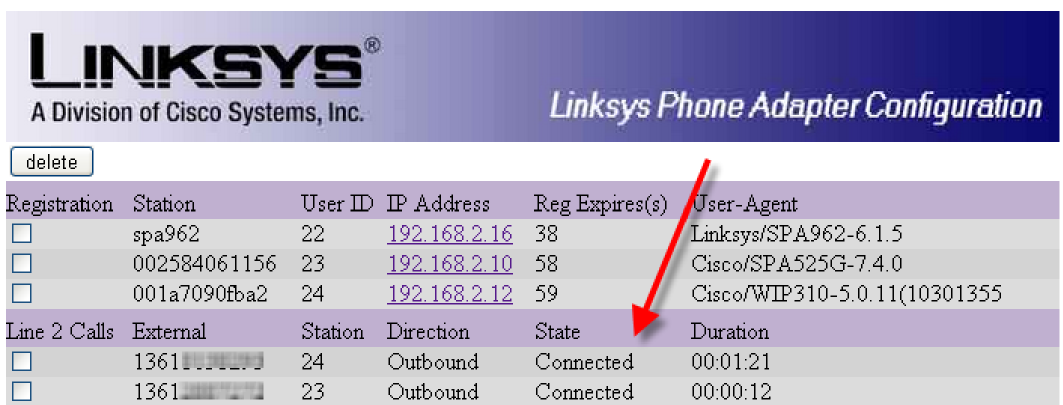
Displaying Useful PBX & Phone Information

Click PBX Status to view:

- Phone name
- Extension number
- IP Address
- Firmware version
- Phone number of other party
- Duration of call



A list of is displayed:



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](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
   SPA9000 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

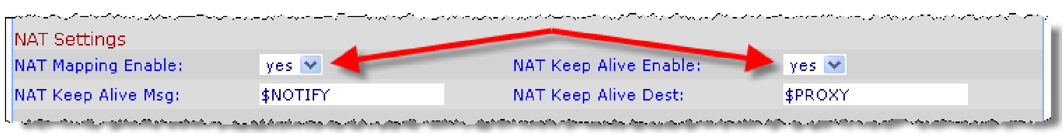
    g. STUN Enable: yes

    h. STUN Server: stun.skype.com

2. Enable NAT
   SPA9000 web-UI > Voice tab > Line *N* > NAT Settings >

    a. NAT Mapping Enable: yes

    b. NAT Keep Alive Enable: yes



2. Click Submit All Changes to save and reboot the SPA9000.

Gathering Information for Support

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

A. SPA9000's configuration: https://www.myciscocommunity.com/docs/DOC-3027

B. SPA5xx's configuration: https://www.myciscocommunity.com/docs/DOC-2982

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