

Configuring the Cisco SPA500 IP Phone with a Skype for SIP Line:



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Introduction

This document assumes that you have a SPA500 IP phone and that you want to configure it 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 just like you would with a regular phone.

The following table provides a summary of the SPA5xx phone family's features:

IP Phone	Line Keys	Color	Attendant Console	WiFi	BlueTooth	PoE	PC Switch Port	Wideband Audio G.722
SPA525G	5	Y	Y	Y	Y	Y	Y	Y
SPA509G	12	Y	Y	N	N	Y	Y	Y
SPA508G	8	N	Y	N	N	Y	Y	Y
SPA504G	4	N	Y	N	N	Y	Y	Y
SPA502G	1	N	Y	N	N	Y	Y	Y
SPA501G	8	n/a	Y	N	N	Y	Y	Y

Audience

This application note is targeted to anyone with a SPA500 series IP phone who wants to save money on phone calls by leveraging Skype for SIP.

Scope

This scope of this document is limited to configuring inbound and outbound call routes on a SPA500 series IP phone and does not address the following topics:

- Security
- Acquiring a Skype for SIP account

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

Related Documents & Resources

- [SPA500 IP phone family data sheets](#)
- [Cisco SPA500 User Guide](#)
- [Cisco SPA500 Series and WIP310 IP Phone Administration Guide](#)
- Cisco Community Central: [Small Business Community IP Phone Support](#)
- [Skype for SIP](#)

Overview

For the purposes of this document, the SPA509G 5-line IP phone with wideband audio support, and 12 lines is used in most examples.

By the end of this document, you will be able to connect and configure a SPA500 series IP phone 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 an extension on the SPA509 IP Phone for Skype for SIP. Extension 1 in this document.
3. Configure the SPA509's name and what must be displayed on the phone's screen, MySPA509G and 3615551212 in this example
4. Configure a name to associate with Line key 1, 5551212 in this example
5. Save changes which causes the SPA500 to reboot and register to Skype for SIP

Requirements

You need the following equipment, services, and information:

- Skype for SIP credentials

Configuring Cisco SPA5xx IP Telephones with a Skype for SIP Line

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

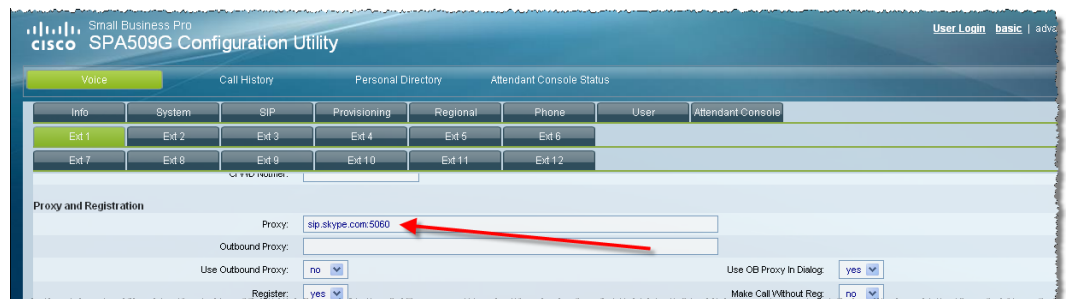
Configuring the SPA500 for Skype for SIP

In this document's example:

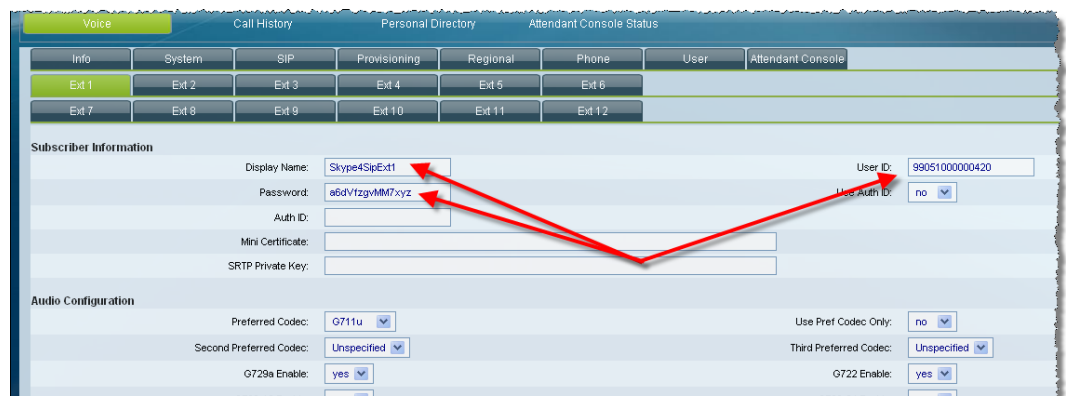
- SPA509 Ext 1 is used for Skype for SIP
- 8 is to be the steering digit [SPA9000 currently uses 9 for SPA400 PSTN calls]
- Inbound calls are routed to the auto attendant (aa)

Configure the SPA509's Ext 1 to register to the Skype for SIP service:

1. Access the SPA509's web user interface (web-UI)
http://<SPA509_IP_address>/admin/advanced
2. Insert **sip.skype.com:5060** into the field at:
SPA509G Voice tab > Ext 1 tab > Proxy and Registration > Proxy:



3. Insert the Skype for SIP user credentials in the
SPA509 Voice tab > Ext 1 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. Configure the SPA509's name at:
SPA509 web-UI Voice tab > Phone tab > General > Station Name: **MySPA509G**

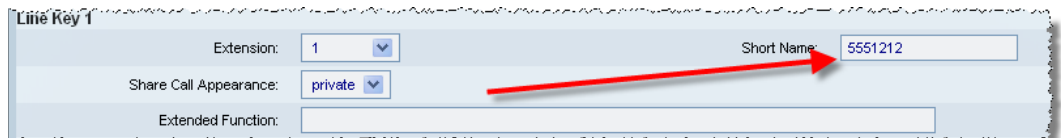
Configuring Cisco SPA5xx IP Telephones with a Skype for SIP Line

- Configure the name to be displayed on the SPA509's display at:
SPA509 web-UI Voice tab > Phone tab > General > Station Display Name: 3615551212



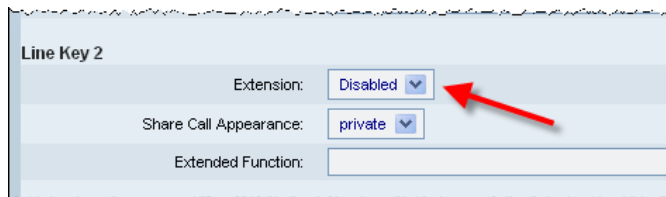
The screenshot shows the 'Voice' tab in the SPA509 web-UI. The 'Phone' sub-tab is selected, and the 'General' section is visible. The 'Station Name' is set to 'MySPA509G' and the 'Station Display Name' is set to '3615551212'. Other fields like 'Voice Mail Number', 'Text Logo', and 'BMP Picture Download URL' are empty. A 'Default' dropdown is visible at the bottom left.

- Configure the name to display for Line Key 1 at
SPA509 web-UI Voice tab > Phone tab > Line Key 1 > Short Name: 5551212



The screenshot shows the 'Line Key 1' configuration page. The 'Extension' is set to '1' and the 'Short Name' is set to '5551212'. A red arrow points to the 'Short Name' field. The 'Share Call Appearance' is set to 'private' and the 'Extended Function' field is empty.

- Disable any unused keys:
SPA509 web-UI Voice tab > Phone tab > Line Key N > Extension: Disabled



The screenshot shows the 'Line Key 2' configuration page. The 'Extension' is set to 'Disabled'. A red arrow points to the 'Disabled' dropdown. The 'Share Call Appearance' is set to 'private' and the 'Extended Function' field is empty.

- Click Submit All Changes. The SPA509 will reboot and be ready for use.

This completes configuring the SPA509 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. If the phone plays a busy signal, this indicates that the phone's dial plan does not permit the dialed numbers. Correct the phone's dial plan. 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 SPA509's Skype for SIP line is registered as follows:

SPA509 web-UI > Info tab > Ext 1 Status > Registration State:



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 SPA500 IP phone 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

SPA509G 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

NAT Support Parameters	
Handle VIA received:	yes
Handle VIA rport:	yes
Insert VIA received:	yes
Insert VIA rport:	yes
Substitute VIA Addr:	yes
Send Resp to Src Port:	yes
STUN Enable:	yes
STUN Test Enable:	no
STUN Server:	stun.skype.com
EXT RTP Port Min:	
EXT IP:	
NAT Keep Alive Intvl:	15

2. Enable NAT

SPA509G web-UI > Voice tab > Line *N* > NAT Settings >

- a. NAT Mapping Enable: yes
- b. NAT Keep Alive Enable: yes

NAT Settings	
NAT Mapping Enable:	yes
NAT Keep Alive Enable:	yes
NAT Keep Alive Msg:	\$NOTIFY
NAT Keep Alive Dest:	\$PROXY

2. Click Submit All Changes to save and reboot the SPA509G.

Gathering Information for Support

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

- A. SPA5xx's configuration: <https://www.myciscocommunity.com/docs/DOC-2982>
- B. WireShark trace to allow the support staff to view network interaction.




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