Cisco SPA525G Configuration with VirtualPBX

If you already have a Cisco SPA525G VoIP-enabled phone, you can set up and configure the phone for use with the VirtualPBX system. This guide was written specifically for the Cisco SPA525G. While other Cisco IP phones may work with our system, this guide may or may not be fully applicable.

1. Set up Account with VirtualPBX
Make sure you have set up an account with VirtualPBX, one that is meant to be used with a self-configured IP phone. In doing this, you will get a DID (phone number) and a password from VirtualPBX.

2. Add a 3rd Party Softphone to your account via the web interface
For these steps with screen shots please see the 3rd Party VoIP Phone Setup Guide
   a. Click on the Admin Tab
   b. Click on VoIP phones
   c. Click on Add Softphone
   d. Select 3rd Party
   e. Copy the credentials for the softphone. You will use this information in Step 6.

3. Plug in Phone
Connect to the network first, then plug in the power
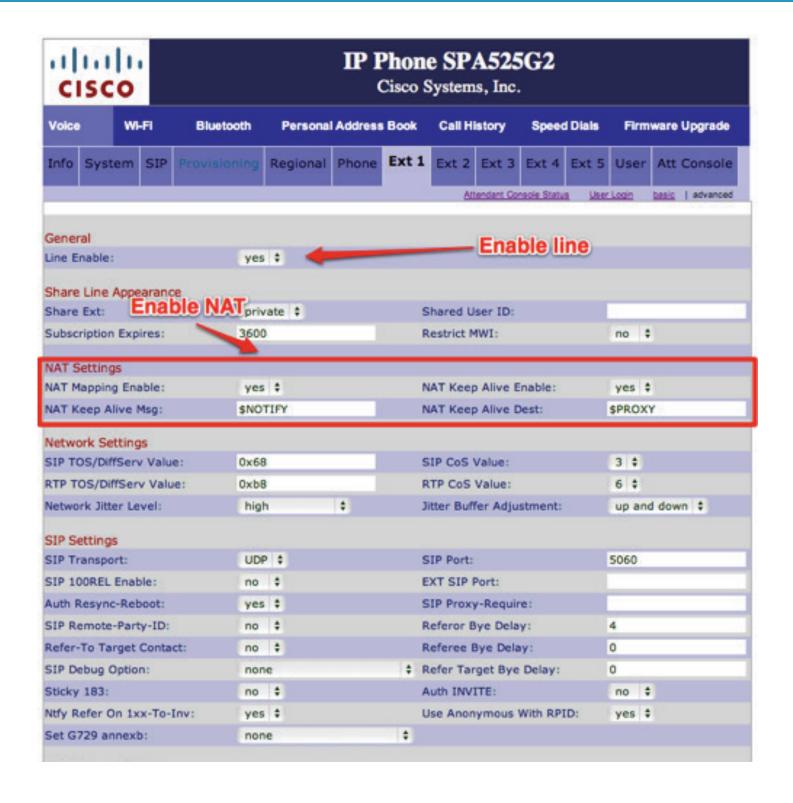
4. Find the Phone's IP Address
   a. On the phone, press the **Setup** button
   b. Select **Option 9 – Network**. Your IP Address should be shown. If it says 0.0.0.0, the device is not properly
     set up on the network and you should talk to your system administrator.

5. Configure Phone via Web Interface
   a. Ensure you are using a computer attached to the same network as the phone
   b. Open a browser window and go to **http://<IP Address>/** (the <IP Address> is the IP address you obtained from
     step #3 above. The format would look like this: **http://192.221.34.51**.)

6. Enter account settings.
   a. Click **Admin Login** on the right side of the page
   b. Click **Advanced** on the right side of the page
   c. Configure your phone as shown on the following page
   d. When finished, click **Submit All Changes** and make sure the phone reboots
   e. The phone should now be ready to use

IP Phone SPA525G2
Cisco Systems, Inc.

Voice | **Wi-Fi** | **Bluetooth** | **Personal Address Book** | **Call History** | **Speed Dials** | **Firmware Upgrade**

Info | System | SIP | Provisioning | Regional | Phone | **Ext 1** | Ext 2 | Ext 3 | Ext 4 | Ext 5 | User | Att Console

Attendant Console Status   User Login   basic | advanced

General

Enable line

Line Enable:	yes

Share Line Appearance

Enable NAT

Share Ext:	private	Shared User ID:	
Subscription Expires:	3600	Restrict MWI:	no

NAT Settings

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

Network Settings

SIP TOS/DiffServ Value:	0x68	SIP CoS Value:	3
RTP TOS/DiffServ Value:	0xb8	RTP CoS Value:	6
Network Jitter Level:	high	Jitter Buffer Adjustment:	up and down

SIP Settings

SIP Transport:	UDP	SIP Port:	5060
SIP 100REL Enable:	no	EXT SIP Port:	
Auth Resync-Reboot:	yes	SIP Proxy-Require:	
SIP Remote-Party-ID:	no	Referor Bye Delay:	4
Refer-To Target Contact:	no	Referee Bye Delay:	0
SIP Debug Option:	none	Refer Target Bye Delay:	0
Sticky 183:	no	Auth INVITE:	no
Ntfy Refer On 1xx-To-Inv:	yes	Use Anonymous With RPID:	yes
Set G729 annexb:	none		

Call Feature Settings

Blind Attn-Xfer Enable:	no	MOH Server:	
Message Waiting:	no	Auth Page:	no
Default Ring:	2	Auth Page Realm:	
Conference Bridge URL:		Auth Page Password:	
Mailbox ID:		Voice Mail Server:	
Voice **Enter Registration Information**		State Ag **Enable DNS SRV**	
CFWD Notify Serv:	no	CFWD Notifier:	

Proxy and Registration

Proxy:	virtualpbx.net		
Outbound Proxy:	virtualpbx.net		
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:	yes	DNS SRV Auto Prefix:	yes
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal

Subscriber Information

Display Name:	357	User ID:	2135993466
Password:	*************	Use Auth ID:	yes
Auth ID:	2135993466		
Mini Certificate:			
SRTP Private Key:			

Audio Configuration

Preferred Codec:	G711u	Use Pref Codec Only:	yes
Second Preferred Codec:	Unspecified	Third P **Enter Subscriber Info**	
G729a Enable:	yes	G722 Enable:	yes
L16 Enable:	yes	G726-32 Enable:	yes
Release Unused Codec:	yes	DTMF Process AVT:	yes
Silence Supp Enable:	no	DTMF Tx Method:	Auto
DTMF Tx Volume for AVT Packet:	0		

Dial Plan

Dial Plan:	(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)
Caller ID Map:	
Enable IP Dialing:	yes	Emergency Number:

Undo All Changes   Submit All Changes

Attendant Console Status   User Login   basic | advanced