Using the 8x8 Registrar: Linksys SPA94x

Linksys SPA94x SIP Phone Configuration

Firmware

The following firmware and handset combinations has been tested and verified:

- Cisco/Linksys/Sipura SPA942 running 6.1.5(a) firmware
- Cisco/Linksys/Sipura SPA941 running 5.1.8 firmware

To obtain the correct release of the Linksys SIP firmware load, navigate to the following URL:

- SPA942 SIP IP Telephones: (Hosted by Cisco Systems)
- SPA941 SIP IP Telephones: (Hosted by Cisco Systems)

This document covers basic and minimum configuration requirements to register a Linksys SPA94x phone to the 8x8 SaaS (Software as a Service) platform. For information on installing firmware or obtaining access to the admin configuration interface, please consult the manufacture's documentation. It is encouraged to get your IT administrator to assist with the configuration of this phone.

SIP Registration

Phones will be configured to register to: vcc-us1-sip.8x8.com by completing this article.

Note: To obtain SIP Proxy for your platform, refer to the Platform URL Guide.

Assumptions

- Administrator has access to handset's web interface.
- Phone has been loaded with appropriate firmware revision.
- Site meets firewall requirements (Outbound UDP:5060 & UDP:35000-65000) to vcc-us1-sip.8x8.com and vcc-us2-sip.8x8.com 's IPs (subject to change).
- Site's Firewall/NAT engines have SIP Application Layer Gateways (ALGs) disabled.
- Handset is set to factory defaults as a reference point.
- Phone is able to obtain an IP address and has network connectivity.

Checking your phone's IP Address

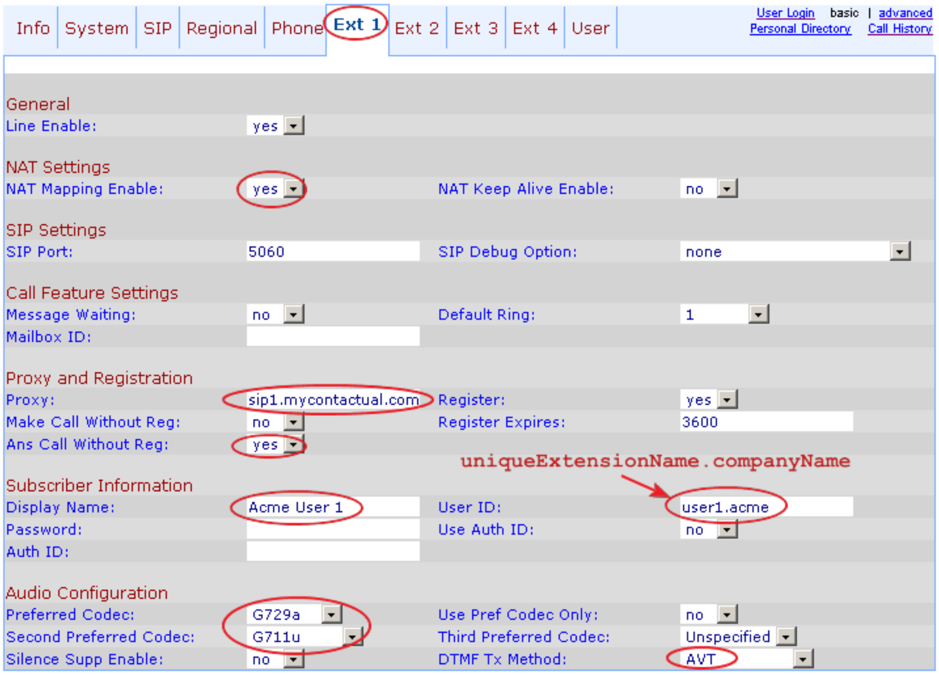
To check your phone's IP Address:

1. Cable your phone to the network and plug in the power supply.

2. Wait for the phone to boot.
3. Click the Setup button (looks like a dog-eared paper, next to the hand).
4. Click or Scroll to option 9 (Network).
5. The second value, labeled "Current IP" is referred to as <ipAddress> in the next step.

Configuration Steps - "Ext1"

1. Using a web browser, navigate to http://<ipAddress>/admin/ to access the phone setup
2. Click **Ext1** tab to edit your Line1 appearance
3. Set the values as below:
    a. *NAT Mapping Enable :* Ask your IT Administrator
        i. YES if behind NAT Firewall/Router
        ii. NO if directly assigned a public IP address to phone
    b. *Proxy:* vcc-us1-sip.8x8.com
    c. *Ans Call Without Reg*: Yes
    d. *Display Name* = <uniqueDisplayName>
    e. *Preferred Codec* = G729a or G711u
    f. *Second Preferred Codec* = G711u or G729a
    g. User ID should contain a unique extension name, which can be any alphanumeric value, followed by a DOT, then a shortened version of your company name to keep values unique between customers (total characters must not exceed (16). The uniqueness of this value is important. Examples are: user1.acme, 1234.dogs, ryan1.siyso, etc. This will be used as the username portion of your SIP URI in the VCC Agent Console and will be entered as "user1.acme@ vcc-us1-sip.8x8.com ", etc.
    h. *DTMF Tx Method* : AVT
4. Click **Submit Changes**. Your phone restarts.

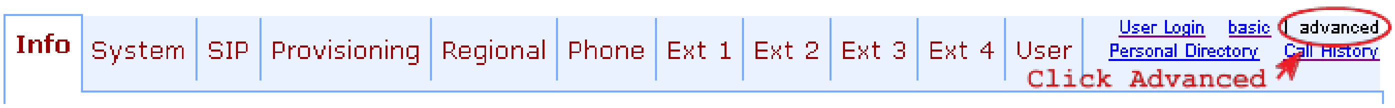

Info
System
SIP
Regional
Phone
Ext 1
Ext 2
Ext 3
Ext 4
User
User Login
basic | advanced
Personal Directory
Call History
General
Line Enable: yes
NAT Settings
NAT Mapping Enable: yes
NAT Keep Alive Enable: no
SIP Settings
SIP Port: 5060
SIP Debug Option: none
Call Feature Settings
Message Waiting: no
Default Ring: 1
Mailbox ID:
Proxy and Registration
Proxy: sip1.mycontactual.com
Register: yes
Make Call Without Reg: no
Register Expires: 3600
Ans Call Without Reg: yes
uniqueExtensionName.companyName
Subscriber Information
Display Name: Acme User 1
User ID: user1.acme
Password:
Use Auth ID: no
Auth ID:
Audio Configuration
Preferred Codec: G729a
Use Pref Codec Only: no
Second Preferred Codec: G711u
Third Preferred Codec: Unspecified
Silence Supp Enable: no
DTMF Tx Method: AVT

* Remember to click "Submit All Changes". Your Phone may restart.

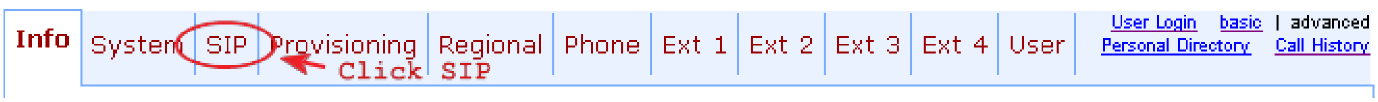
Notes: Do not enable silence suppression. Phones or networks with this setting enabled may be banned.

Configuration Steps - SIP (Advanced)

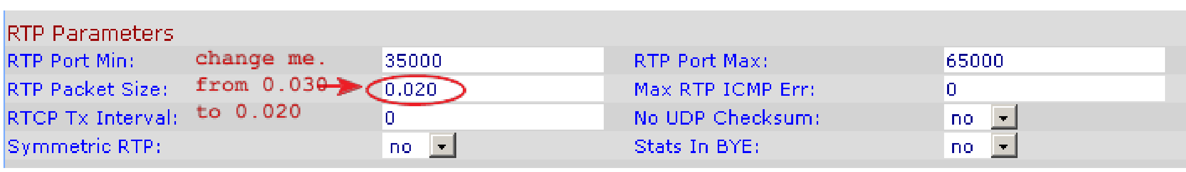
1. Click the "Advanced" text link in the top-right corner of the admin interface.



2. Navigate to the "SIP" tab.



3. Scroll down to "RTP Parameters"
4. Locate and change "RTP Packet Size" from "0.030" to a new value of "0.020".
5. Click **Submit All Changes**. Your Phone may restart.



* Remember to click "Submit All Changes". Your Phone may restart.

Virtual Contact Center Agent Console Settings

1. Login to the Agent Console as a regular user
2. Click **Offline** if necessary.
3. Click **My Profile** on the top toolbar.
4. Set the SIP Phone URI based on the "User ID" defined earlier, plus "@ vcc-us1-sip.8x8.com ". Our example throughout this document has been "user1.acme@ vcc-us1-sip.8x8.com ". Define the value appropriate to your handset's configuration.
5. Click **Save.**
6. Click "Make Verification Call" and you should hear a pre-recorded message stating "This call allows you to test your phone connection.