

## Cisco SPA122 Configuration Guide for FreePhoneLine Software Version: 1.3.3(015)

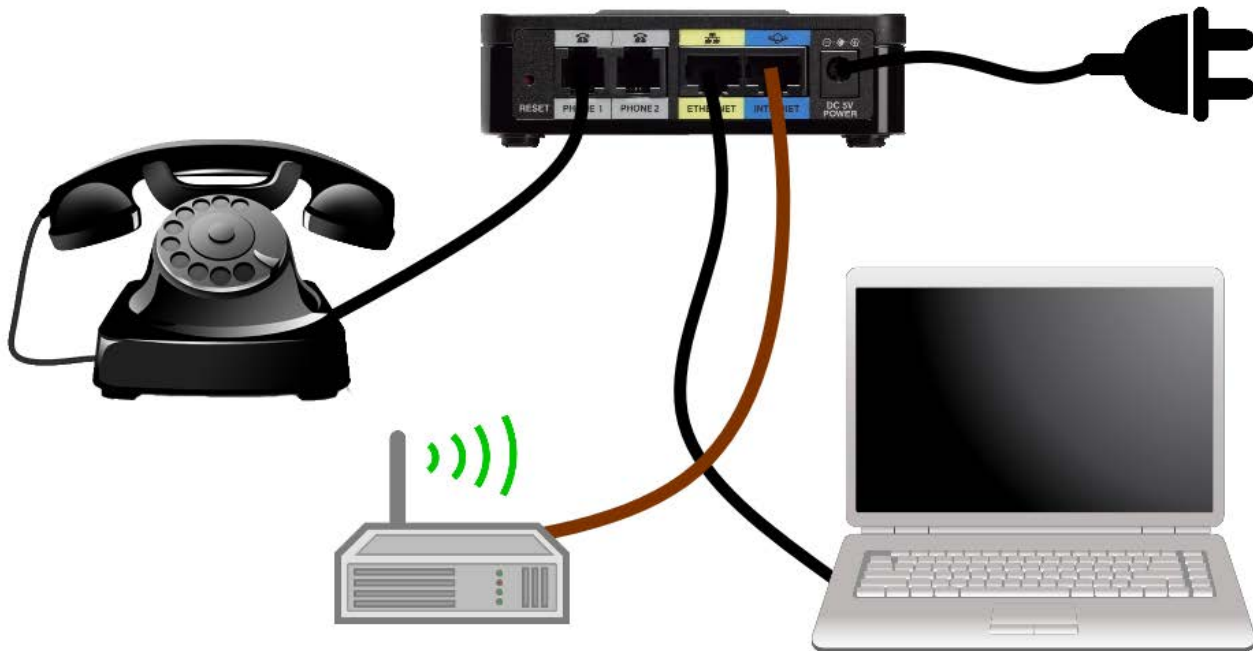
Official FPL guidelines you can find at: <http://support.freephoneline.ca/entries/23120323-VoIP-Unlock-Key-Credentials>

**All the settings were default on the device before configuration.**

### Step 1.

Plug all the appropriate wires to your Linksys SPA122 device.

- Ethernet cable *[Note: Connect to Ethernet Port on your SPA122 device and another end to your PC/laptop]*
- Phone line (attached to the phone) *[Note: Use 'Phone 1' port]*
- Power



### Step 2.

On the PC/Mac go to your browser (Internet Explorer; Chrome; Firefox; Opera; Safari, etc.).

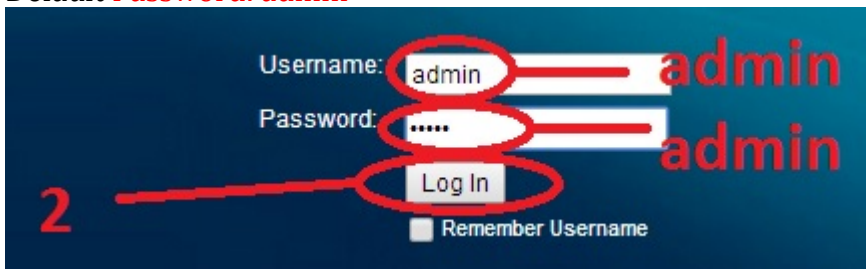
In the URL bar put IP address: **192.168.15.1**



Login page will appear.

Default **Username: admin**

Default **Password: admin**



Step 3.

Go to the 'Network Setup' tab, then on your left side menu click on 'Time Settings' and change the following settings:

Under 'Time Zone' Category choose the appropriate time zone where SPA122 is located.

Then click 'Submit' button to save changes.

Wait until changes are saved.

Quick Setup **Network Setup** Voice Administration Status

Basic Setup  
Internet Settings  
**Time Settings**  
Advanced Settings

Time Settings

☐ User Manual

Date: 2014 / 7 / 16 (Year/Month/Day)

Time: 13 : 16 : 27 (Hour:Min:Sec)

☒ Time Zone

(GMT-05:00) Eastern Time (USA & Canada)

☒ Adjust Clock for Daylight Saving Changes

Time Server: Manual 0.ciscosb.pool.ntp.org

Resync Timer: 3600 seconds

Auto Recovery After Reboot: ☐

Submit Cancel

(GMT-11:00) Midway Island, Samoa  
(GMT-10:00) Hawaii  
(GMT-09:00) Alaska  
(GMT-08:00) Pacific Time (USA & Canada)  
(GMT-07:00) Arizona  
(GMT-07:00) Mountain Time (USA & Canada)  
(GMT-06:00) Mexico  
(GMT-06:00) Central Time (USA & Canada)  
(GMT-05:00) Indiana East, Colombia, Panama  
(GMT-05:00) Eastern Time (USA & Canada)  
(GMT-04:00) Bolivia, Venezuela  
(GMT-04:00) Atlantic Time (Canada), Brazil West  
(GMT-04:00) Guyana  
(GMT-03:30) Newfoundland  
(GMT-03:00) Brazil East, Greenland  
(GMT-02:00) Mid-Atlantic  
(GMT-01:00) Azores  
(GMT) Gambia, Liberia, Morocco  
(GMT) England  
(GMT+01:00) Tunisia

Step 4.

Go to the 'Voice' tab, then on your left side menu go to 'SIP' tile and change the following settings:

Under 'RTP Parameters' category

**RTP Packet Size: 0.020**

Under 'NAT Support Parameters' category

**NAT Keep Alive Intvl: 20**

Then click 'Submit' button to save changes.

Wait until changes are saved.

Quick Setup Network Setup **Voice** Administration Status

Information System SIP Provisioning Regional Line 1 User 1 Line 2 User 2

### SIP

**RTP Parameters**

RTP Port Min: 16384 RTP Port Max: 16482

RTP Packet Size: 0.020 RTP Tx Packet Size Follows Remote SDP: yes

Max RTP ICMP Err: 0 RTCP Tx Interval: 0

No UDP Checksum: no Stats In BYE: yes

**SDP Payload Types**

NSE Dynamic Payload: 100 AVT Dynamic Payload: 101

INFOREQ Dynamic Payload: G726r32 Dynamic Payload: 2

G729b Dynamic Payload: 99 EncapRTP Dynamic Payload: 112

RTP-Start-Loopback Dynamic Payload: 113 RTP-Start-Loopback Codec: G711u

NSE Codec Name: AVT Codec Name: telephone-event

G711u Codec Name: PCMA

G726r32 Codec Name: G726-32

G729b Codec Name: G729ab

G729a Codec Name: G729a

EncapRTP Codec Name: encaprtcp

**NAT Support Parameters**

Handle VIA received: no

Insert VIA received: no

Substitute VIA Addr: no

STUN Enable: no

STUN Server: EXT RTP Port Min: NAT Keep Alive Intvl: 20

Handle VIA rport: no

Insert VIA rport: no

Send Resp To Src Port: no

STUN Test Enable: no

EXT IP: NAT Keep Alive Intvl: 20

Submit Cancel Refresh

### Step 5.

Go to 'Voice' tab, then go 'Regional' line and change the following settings:

Under 'Ring and Call Waiting Tone Spec' category:

**Ring Waveform:** *Sinusoid*

**Ring Frequency:** 52

**Ring Voltage:** 90

Then click 'Submit' button to save changes.

Wait until changes are saved.

Quick Setup Network Setup **Voice** Administration Status

Information System SIP Provisioning **Regional** Line 1 User 1 Line 2 User 2

### Regional

Ring7 Name: Belcore-r7 Ring8 Name: Belcore-r8

**Ring and Call Waiting Tone Spec**

Ring Waveform: Sinusoid Ring Frequency: 52

Ring Voltage: 90 CWT Frequency: 440@-10

Synchronized Ring: no

**Control Timer Values (sec)**

Hook Flash Timer Min: .1 Hook Flash Timer Max: .9

Callee On Hook Delay: 0 Reorder Delay: 5

Call Back Expires: 1800 Call Back Retry Intvl: 30

Call Back Delay: .5 VMWI Refresh Intvl: 0

Submit Cancel Refresh

**Step 6.**

Go to 'Voice' tab, then go 'Line 1' line and change the following settings:

Under 'NAT Settings' category:

**NAT Mapping Enable:** *yes*

**NAT Keep Alive Enable:** *yes*

Quick Setup Network Setup **Voice** Administration Status

Information System SIP Provisioning Regional **Line 1** User 1 Line 2 User 2

**Line 1**

**General**  
Line Enable:

**Streaming Audio Server (SAS)**  
SAS Enable:  SAS DLG Refresh Intvl:   
SAS Inbound RTP Sink:

**NAT Settings**  
NAT Mapping Enable:  NAT Keep Alive Enable:   
NAT Keep Alive Msg:  NAT Keep Alive Dest:

Under 'Proxy and Registration' category:

**Proxy:** *voip.freephoneline.ca* **OR** *voip2.freephoneline.ca* [Note: For **ROGERS** Internet provider customers use *voip4.freephoneline.ca:6060*]

**Register Expires:** *3600*

Under 'Subscriber Information' category:

**Display Name:** *[Your first and last name]*

[Note: ATA and SIP clients with a Caller ID string containing non-alphanumeric characters will prevent you from making outgoing calls]

**User ID:** *[Your FPL number 1xxxxxxxxx]*

**Password:** *[Your SIP password]*

**Line 1**

**Proxy and Registration**  
Proxy:   
Outbound Proxy:   
Use Outbound Proxy:   
Register:   
Register Expires:   
Use DNS SRV:   
Proxy Fallback Intvl:   
Mailbox Subscribe URL:   
Use OB Proxy In Dialog:   
Make Call Without Reg:   
Ans Call Without Reg:   
DNS SRV Auto Prefix:   
Proxy Redundancy Method:   
Mailbox Subscribe Expires:

**Subscriber Information**  
Display Name:   
Password:   
Auth ID:   
SIP URI:   
User ID:   
Use Auth ID:   
Resident Online Number:

Under 'Audio Configuration' category:

**Preferred Codec:** *G711u*

**Second Preferred Codec:** *G729a*

**Third Preferred Codec:** *G711u*

Under 'Dial Plan' category:

**Dial Plan:** *(911|[2-9]xxxxxxxx/1xxxxxxxx/011xxxxxxxxxx/98\*|[6-7]x\*xxxxxxxxxx.)*

Then click 'Submit' button to save changes.

Wait until changes are saved.

Line 1

Reuse CID Number As Name:  CONF CID Serv:

**Audio Configuration**

Preferred Codec:  Second Preferred Codec:

Third Preferred Codec:

Use Remote Pref Codec:

G729a Enable:

G726-32 Enable:

FAX V21 Detect Enable:

FAX CNG Detect Enable:

FAX Codec Symmetric:

FAX Passthru Method:

FAX Process NSE:

FAX Disable ECAN:

DTMF Tx Strict Hold Off Time:

Hook Flash Tx Method:

FAX T38 ECM Enable:

Symmetric RTP:

Modem Line:

Use Pref Codec Only:

Codec Negotiation:

Silence Supp Enable:

Silence Threshold:

Echo Canc Enable:

FAX Passthru Codec:

DTMF Process INFO:

DTMF Process AVT:

DTMF Tx Method:

DTMF Tx Mode:

FAX Enable T38:

FAX T38 Redundancy:

FAX Tone Detect Mode:

FAX T38 Return to Voice:

RTP to Proxy in Remote Hold:

**Dial Plan**

Dial Plan:

### Step 8.

Go to 'Voice' tab, then click on 'Line 2':

**Line Enable:** *no*

Quick Setup Network Setup **Voice** Administration Status

Information System SIP Provisioning Regional Line 1 User 1 **Line 2** User 2

Line 2

**General**

Line Enable:

**Streaming Audio Server (SAS)**

SAS Enable:

SAS Inbound RTP Sink:

SAS DLG Refresh Intvl:

Then click 'Submit' button at the bottom.

Wait until changes are saved.

Enjoy your free phone line! 😊