

Cisco SPA112 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 CISCO SPA112 device.

- Ethernet cable (Internet connection)
- Phone line (attached to the phone) *[Note: Use 'Phone 1' port]*
- Power

Step 2.

To find out IP address for accessing web-based utility from your phone attached to SPA112 dial:

- '****'
- Then dial '110#'

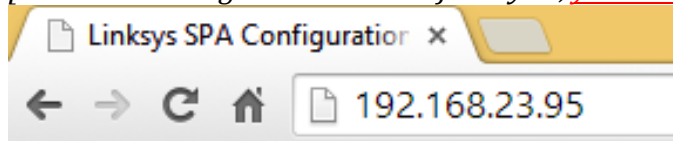
Write down your IP address.

Step 3.

On the PC/Mac connected to the same network as your CISCO SPA112 device, go to your browser (Internet Explorer; Chrome; Firefox; Opera; Safari, etc.).

In the URL bar put your IP address from Step 2.

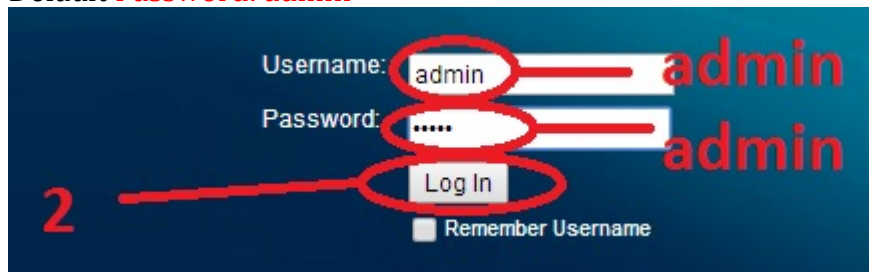
[Note: I am using 192.168.23.95 for my IP, your IP address might be different!]



Login page will appear.

Default **Username: admin**

Default **Password: admin**



Step 4.

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 SPA112 is located.

Then click 'Submit' button to save changes.

Wait until changes are saved.

The screenshot shows the 'Time Settings' configuration page. The 'Network Setup' tab is selected at the top. In the left sidebar, 'Time Settings' is highlighted under the 'Basic Setup' category. The main content area shows the 'Time Zone' section with a dropdown menu set to '(GMT-05:00) Eastern Time (USA & Canada)'. A list of time zones is displayed on the right, with '(GMT-11:00) Midway Island, Samoa' at the top. The 'Submit' button is circled at the bottom.

Annotations:

- 1: Points to the 'Network Setup' tab.
- 2: Points to the 'Time Zone' dropdown menu.
- 3: Points to the 'Submit' button.

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 5.

Go to the 'Voice' tab, then on your left side menu go to 'SIP' line and fill in 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: NSE AVT Codec Name: telephone-event

G711u Codec Name: PCMA G711a Codec Name: PCMA

G726r32 Codec Name: G726-32 G729a Codec Name: G729a

G729b Codec Name: G729ab EncapRTP Codec Name: encaprtcp

NAT Support Parameters

Handle VIA received: no Insert VIA port: no

Substitute VIA Addr: no Send Resp To Src Port: no

STUN Enable: no STUN Test Enable: no

STUN Server: EXT IP: NAT Keep Alive Intvl: 20

EXT RTP Port Min: Submit Cancel Refresh

Step 6.

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: Bellcore-r7 Ring8 Name: Bellcore-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 7.

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

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

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: ☐ yes CONF CID Serv: ☐ yes

Audio Configuration

Preferred Codec: 4

Third Preferred Codec: 4

Use Remote Pref Codec: ☐ no

G729a Enable: ☐ yes

G726-32 Enable: ☐ yes

FAX V21 Detect Enable: ☐ yes

FAX CNG Detect Enable: ☐ yes

FAX Codec Symmetric: ☐ yes

FAX Passthru Method:

FAX Process NSE: ☐ yes

FAX Disable ECAN: ☐ no

DTMF Tx Strict Hold Off Time:

Hook Flash Tx Method:

FAX T38 ECM Enable: ☐ yes

Symmetric RTP: ☐ no

Modem Line: ☐ no

Second Preferred Codec: 4

Use Pref Codec Only: ☐ no

Codec Negotiation:

Silence Supp Enable: ☐ no

Silence Threshold:

Echo Canc Enable: ☐ yes

FAX Passthru Codec:

DTMF Process INFO: ☐ yes

DTMF Process AVT: ☐ yes

DTMF Tx Method:

DTMF Tx Mode:

FAX Enable T38: ☐ no

FAX T38 Redundancy:

FAX Tone Detect Mode:

FAX T38 Return to Voice: ☐ no

RTP to Proxy in Remote Hold: ☐ no

Dial Plan

Dial Plan: 5

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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: 2

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! ☺